DEFENSE INFORMATION SYSTEMS AGENCY
P. O. BOX 4502
ARLINGTON, VIRGINIA 22204-4502

IN REPLY REFER TO: Joint Interoperability Test Command (JTE) **13 Aug 09**

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Special Interoperability Test Certification of PacStar 6800 Enterprise Unified Capabilities (UC) Exchange with software version IQ-Core 3.0

References: (a) DoD Directive 4630.5, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) CJCSI 6212.01D, "Interoperability and Supportability of Information Technology and National Security Systems," 8 March 2006
(c) through (f), see Enclosure 1

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.

2. The PacStar 6800 Enterprise UC Exchange with software version IQ-Core 3.0 is hereinafter referred to as the System Under Test (SUT). The SUT, which includes a Cisco enclave and a REDCOM enclave, met all of its critical interoperability requirements and is certified as interoperable for joint use within the DSN. The SUT was tested and met the critical interoperability requirements for the following DSN switch types: Small End Office (SMEO), Private Branch Exchange (PBX) 1, and PBX 2. The SUT is certified for Voice over Internet Protocol (VoIP) with certified Assured Services Local Area Networks (ASLANs) on the UC Approved Products List (APL). The listed test discrepancies shown in the Certification Testing Summary (Enclosure 2) have an overall minor operational impact. No other configurations, features, or functions, except those cited within this report, are certified by the JITC. This certification expires upon changes that could affect interoperability, but no later than three years from the date of this memorandum.

3. This finding is based on interoperability testing conducted by JITC, review of the vendor's Letters of Compliance (LoC), and Defense Information Assurance (IA)/Security Accreditation Working Group (DSAWG) accreditation. Testing was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 2 June through 25 July 2008. Regression testing was conducted from 9 January through 13 February 2009 to test configuration changes and patches developed to fix test discrepancies discovered during initial testing. Review of the vendor's LoC was completed on 8 April 2009. DSAWG grants accreditation based on the security testing completed by DISA-led Information Assurance test teams and published in a separate report (reference (c)). DSAWG accreditation was granted on 11 August 2009.

JITC Memo, JTE, Special Interoperability Test Certification of PacStar 6800 Enterprise Unified Capabilities (UC) Exchange with software version IQ-Core 3.0

Enclosure 2 documents the test results and describes the tested network and system configurations

.

4. The interoperability test summary of the SUT is contained in Table 1. The SMEO required and conditional Capability Requirements (CRs) and Feature Requirements (FRs) are listed in Table 2. This interoperability test status is based on the SUT's ability to meet:

a. DSN services for Network and Applications specified in reference (d).

b. SMEO interface and signaling requirements for trunks/lines specified in reference (e) verified through JITC testing and/or vendor submission of LoC.

c. SMEO CRs/FRs specified in reference (e) verified through JITC testing and/or vendor submission of LoC.

d. The overall system interoperability performance derived from test procedures listed in reference (f).

Table 1. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, DP)	Yes	Certified	Met all critical CRs and FRs with the REDCOM enclave. Although Cisco offers this interface, it is not certified.
T1 CAS (MFR1)	No	Certified	Met all critical CRs and FRs with the REDCOM enclave. Although Cisco offers this interface, it is not certified.
E1 CAS (DTMF, DP)	Yes (Europe only)	Certified	Met all critical CRs and FRs with the REDCOM enclave. Although Cisco offers this interface, it is not certified.
E1 CAS (MFR1)	No (Europe only)	Certified	Met all critical CRs and FRs with the REDCOM enclave. Although Cisco offers this interface, it is not certified.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the REDCOM enclave. The Cisco enclave does not support NFAS.[1] The Cisco enclave monitoring tool occasionally provides inaccurate reports when a remote trunk is busy.[2]
E1 PRI (ITU-T Q.955.3)	No (Europe only)	Certified	Met all critical CRs and FRs with the REDCOM enclave. The E1 ISDN PRI interface is supported by the Cisco enclave; however, it does not support ITU-T Q.955.3 MLPP.
T1 SS7 (ANSI T1.619a)	No	Not Tested	T1 SS7 is supported by the SUT; however it was not tested. The SUT T1 SS7 interface is therefore not certified by JITC. This is not a required interface for a SMEO.
E1 SS7 (ANSI T1.619a)	No	Not Tested	E1 SS7 is not supported by the SUT. This is not a required interface for a SMEO. There is no risk associated with the SUT not supporting this interface.
2-wire/4-wire E&M	No	Not Tested	2-wire/4-wire E&M is supported by the SUT; however it was not tested. The SUT 2-wire/4-wire E&M interface is therefore not certified by JITC. This is not a required interface for a SMEO.
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs with a minor configuration change[3] and the following minor exceptions: The REDCOM enclave conference disconnect tone on phones connected to the HDX switch do not meet the specifications.[4]
ISDN BRI NI 1/2	Yes	Certified	Met all critical CRs and FRs with the REDCOM enclave with the following minor exceptions: The conference disconnect tone does not meet the specifications.[4] The precedence above ROUTINE ringing cadence that the SUT applies to BRI phones does not meet the specifications.[5] This interface is not supported on the Cisco enclave.

Table 1. SUT Interoperability Test Summary (continued)

DSN Line Interfaces				
Interface & Signaling		**Critical**	**Status**	**Remarks**
2-Wire Proprietary Digital		No	Not Tested	2-Wire Proprietary Digital is not supported by the SUT. This is not a required interface for a SMEO. There is no risk associated with the SUT not supporting this interface.
VoIP (Session Initiation Protocol)		No	Certified	Met all critical CRs and FRs with the Cisco enclave.
Common Features		Yes	Certified	Met all critical CRs and FRs for the with the following minor exception: The SUT does not support Call Pickup between the two enclaves.[6] The REDCOM enclave does not correctly support the call forwarding variable "ping" ring feature.[7] Met all critical CRs and FRs for the Cisco enclave with the following minor exceptions: Full compliance of DSN Common Call Features was not met.[8, 9, 10, 11, 12, 13]
Attendant		No	Not Tested	The SUT does not support this feature. This is not a required feature for a SMEO. There is no risk associated with the SUT not supporting this feature.
Public Safety		Yes	Certified	Met all critical CRs and FRs.
Conferencing	Preset	No	Certified	Met all critical CRs and FRs with the REDCOM enclave.
	Meet-me	Yes	Certified	Met all critical CRs and FRs with the REDCOM enclave.
	Progressive	No	Certified	Met all critical CRs and FRs with the REDCOM enclave.
Nailed-up Connections		No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a SMEO. There is no risk associated with the SUT not supporting this feature.
DSN Hotline Services		Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT does not support the ISDN PRI code set 5 off hook indicator elements for hotline services.[14]
MLPP		Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support the loss of Command and Control announcement.[15] The SUT does not support Method 1 preemption search algorithim if the trunks are a combination of Cisco and REDCOM enclaves.[16] The SUT does not maintain the precedence level when transferring a call from the Cisco enclave to the REDCOM enclave.[17] The SUT does not support Call Pickup between the Cisco enclave and the REDCOM enclave.[6] When the initiator of a three-way call is preempted, the remaining parties do not receive a conference disconnect tone.[18]
Call Processing		Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The REDCOM enclave does not support the full complement of CoS tables.[19] The SUT does not support calling number delivery.[20]
Network Management		Yes	Certified	Met all critical CRs and FRs with Internet Protocol (IP) interfaces.
ISDN Services		Yes	Certified	Met all critical CRs and FRs. The Cisco enclave does not support NFAS. NFAS is supported on the REDCOM enclave.[1] The operational impact is minor.
Synchronization		Yes	Certified	Met all critical CRs and FRs.
Reliability		Yes	Certified	Met all critical CRs and FRs.[21]
Security		Yes	Certified	See note 22.
VoIP System		No	Certified	The SUT is certified for VoIP with any certified ASLAN posted on the UC APL. See notes 23 and 24.
Network Gateways				
Gateway	**Interface & Signaling**	**Critical**	**Status**	**Remarks**
PSTN	T1 CAS (DTMF, DP, MFR1)	Yes	Certified	Met all critical CRs and FRs with the REDCOM enclave. Although Cisco offers this interface, it is not certified.
	E1 CAS (DTMF, DP, MFR1)	Yes (Europe only)	Certified	Met all critical CRs and FRs with the REDCOM enclave. Although Cisco offers this interface, it is not certified.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	Yes	Certified	Met all critical CRs and FRs with the REDCOM enclave. The Cisco enclave does not support NFAS.[1] The operational impact is minor.
	E1 PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.
	Ground Start Line	Yes	Certified	Met all critical CRs and FRs.

Table 1. SUT Interoperability Test Summary (continued)

NOTES:

1 The SUT does not support NFAS on the Cisco enclave ISDN PRI National ISDN Standard 2 (NI2) interface. NFAS is supported on the REDCOM HDX. Therefore, the REDCOM enclave must include at least one HDX shelf equipped with multiple ISDN PRI T1s. Both enclaves do support FAS.

2 A discrepancy exists that is associated with the monitoring tool that Cisco enclave uses to check the status of the ISDN PRI trunks on the gateway. The monitoring tool occasionally provides an inaccurate representation of the status of the channels on the trunks when they are busied by the remote switching system. The SUT will occasionally provide an indication that the channel that was busied out by the far-end switch remains in an idle condition. This anomaly can be eliminated by insuring the trunks are busied at both the remote end and at the SUT. Furthermore, when this anomaly does occur, the correct busy state of the trunks is reflected in layer 3 protocol of the ISDN PRI interface, therefore, the operational impact is minor.

3 A configuration change was required on the Cisco enclave analog gateways to meet the requirement for interoperability with secure devices, specifically the L3 Omni Secure Wireline Terminal. On the individual voice ports, the minimum and maximum settings for "timing hookflash in" had to be changed to a maximum value of 500 ms and a minimum value of 150 ms. Otherwise, a call that is placed between two Omni devices on the SUT will not disconnect when placed on hook.

4 The conference disconnect tone that is provided by the REDCOM enclave does not meet the specifications designated in UCR, section 5.5.2. The SUT conference disconnect tone is distinguishable from other DSN tones and cadences; therefore, this anomaly has a minor operational impact.

5 The precedence above ROUTINE ringing cadence that the REDCOM enclave applies to BRI phones does not meet the specifications as detailed in the UCR, section 5.5.1. The precedence above ROUTINE cadence is distinct from the ROUTINE cadence when it is configured properly; therefore this anomaly has no operational impact.

6 The SUT does not support Call Pickup between the Cisco enclave and the REDCOM enclave. This solution is unique in that it offers SMEO functionality with two switches (Cisco and REDCOM) and each switch offers call pickup. This action is mitigated by not mixing call pickup groups between the two switch enclaves. DISA's adjudication of this discrepancy was ruled to have a minor operational impact.

7 When CFV is assigned to any station on the REDCOM enclave and CFV is invoked by the user, any station with CFV invoked does not receive a "ping" ring when calls are being forwarded. The operational impact is minor.

8 Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog stations of the Cisco enclave. Call Forward Variable, Three-way Calling, Call Hold, and Call Transfer are supported on VoIP stations only. These features are required for a SMEO for all instruments; however, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. Denied Originating Service is not supported by the SUT and is therefore not covered in this certification. This feature is not required for a SMEO.

9 The Cisco enclave does not support Call Waiting. However, there is no operational impact because the requirement is satisfied with multiple line appearances having a busy trigger. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

10 All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions.

11 Although the Cisco enclave does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP stations. This provides the ability for a user to receive additional calls while active with another call. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. There is no operational impact.

12 A short "ping" ring is not provided when calls are forwarded on the Cisco enclave; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact.

13 When a ROUTINE call is placed to a hunt group, and a ring-no-answer condition occurs, the calling party is diverted to the MLPP alternate directory number. This configuration must be done to allow correct treatment to be provided to precedence calls above ROUTINE that are placed to the hunt group. The UCR requires this only for precedence above ROUTINE calls. There is no operational impact.

14 The SUT does not support the ISDN PRI code set 5 off hook indicator elements for hotline services as required by the UCR. The vendor began testing prior to 14 June 2008 and, therefore, was not required to provide this feature. This anomaly has minor operational impact.

15 The SUT does not support the Loss of C2 announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. This requirement is currently under review by DISA and the Joint Staff. In addition, the specific conditions that invoke this announcement have not yet been defined. As a result, the vendors are not required to be in compliance until 18 months from the date the requirement is fully defined.

16 The SUT does not support Method 1 preemption search algorithm if the trunks are a combination of the Cisco and REDCOM enclaves. In order to use the Method 1 search preemption search algorithm, all trunk groups must be member of the Cisco Gateway or the HDX switch. DISA's adjudication of this discrepancy was ruled to have a minor operational impact.

17 The SUT does not maintain the precedence level when transferring a call between the Cisco enclave and the REDCOM enclave. This discrepancy is due to the functionality between the Cisco and REDCOM enclaves. DISA's adjudication of this discrepancy was ruled to have a minor operational impact.

18 When the initiator of a three-way call is preempted, the remaining parties do not receive a conference disconnect tone. However, the remaining members of the three-way call do stay connected. DISA's adjudication of this discrepancy was ruled to have a minor operational impact.

Table 1. SUT Interoperability Test Summary (continued)

NOTES (continued):

19 The SUT does not support the full complement of CoS tables as specified in the UCR. The SUT supports 255 CoS tables for analog lines and does not support CoS tables on access lines, number codes, trunks, or groups of trunks. This limitation has posed a minor operational impact within the DSN when assigning lines and trunks on the SUT.

20 This is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

21 Backup power, power components, UPS requirements, UPS load capacity and alarms are non-testable requirements. It is the responsibility of the respective base/post/camp/station communication agency to provide this with the SUT when installed.

22 Security is tested by DISA-led Information Assurance test teams and published in a separate report, reference (c).

23 An IPv6 capable system or product, as defined in the UCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor LoC signed by the Vice President of their respective company. The vendor stated in writing, their intent to return to JITC for testing of their solution with IPv6 enabled earliest date available. In addition they stated in writing, compliance to the following criteria:

a. Conformant with IPv6 standards profile contained in the Department of Defense Information Technology Standards Registry (DISR). These standards are delineated in the UCR, appendix 11.

b. Maintaining interoperability in heterogeneous environments and with IPv4.

c. Commitment to upgrade as the IPv6 standard evolves.

d. Availability of contractor/vendor IPv6 technical support.

24 The SUT was tested with IPv4 only. In accordance with the Office of Secretary IPv6 Rules of engagement a solution can be tested and certified for IPv4 only, however the vendor is required to stipulate in an IPv6 LoC their way ahead to be IPv6 capable by end of CY 2008. In addition the vendor is required to return for retest with this IPv6 solution prior to the end of CY 2008. The vendor stated in their IPv6 LoC submission that they will not be able to deliver an IPv6 capable solution until 31 May of 2010. The vendor received a waiver for this requirement from OSD on 9 March 2009.

LEGEND:

ANSI	American National Standards Institute	GR-506-CORE	LSSGR: Signaling for Analog Interfaces	OSD	Office of the Secretary of Defense
APL	Approved Products List	HDX	High Density Exchange	PRI	Primary Rate Interface
ASLAN	Assured Services Local Area Network	IPv4	Internet Protocol version 4	PSTN	Public Switched Telephone Network
BRI	Basic Rate Interface	IPv6	Internet Protocol version 6	Q.931	Signaling Standard for ISDN
C2	Command and Control	ISDN	Integrated Services Digital Network	Q.955.3	ISDN signaling standard for E1 MLPP
CAS	Channel Associated Signaling	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	SMEO	Small End Office
CFV	Call Forwarding Variable			SS7	Signaling System 7
CoS	Class of Service			SUT	System Under Test
CRs	Capability Requirements	JITC	Joint Interoperability Test Command	T1	Digital Transmission Link Level 1 (1.544 Mbps)
CY	Calendar Year	LoC	Letters of Compliance	T1.607	ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
DISA	Defense Information Systems Agency	LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements		
DP	Dial Pulse				
DSN	Defense Switched Network				
DSS1	Digital Subscriber Signaling 1	Mbps	Megabits per second		
DTMF	Dual Tone Multi-Frequency	MFR1	Multi-Frequency Recommendation 1	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
E&M	Ear and Mouth				
E1	European Basic Multiplex Rate (2.048 Mbps)	MLPP	Multi-Level Precedence and Preemption	UC	Unified Capabilities
				UCR	Unified Capabilities Requirements
FAS	Facility Associated Signaling	ms	milliseconds		
FRs	Feature Requirements	NFAS	Non-Facility Associated Signaling	UPS	Uninterruptible Power Supply
GR	Generic Requirement	NI 1/2	National ISDN Standard 1 or 2	VoIP	Voice over Internet Protocol

Table 2. SMEO Requirements

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 SS7 (ANSI T1.619a) E1 SS7 (ITU-T Q.735.3) T1 CAS (MFR1) T1 CAS (DTMF, DP) E1 CAS (DTMF, DP) E1 CAS (MFR1) T1 ISDN PRI NI 1/2 (ANSI T1.619a) E1 ISDN PRI (ITU-T Q.955.3)	No No (Europe only) No Yes Yes (Europe only) No (Europe only) Yes No (Europe Only)	Trunking	• Direct Inward Dialing (C) • National ISDN 1/2 Primary Access (R) • ISDN ANSI MLPP Service Capability (R) • ITU-T ISDN Primary Access (Europe only) (C) • ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (Europe only) (C) • Normal Wink Start Operations (R) • Glare Operation (R) • Abnormal Wink Start (R) • Glare Resolution (R) • Call for Service Timing (R) • Guard Timing (R) • Satellite Timing (R) • Disconnect Control (R) • Reselect and Retrial (R) • Off-Hook Supervision Transition (R) • Dial-Pulse Signals (R) • DTMF Signaling (R) • Standard Digit Format for Precedence (C) • MFR1 2/6 Signaling (C) • Alerting Signals and Tones (R) • Common Channel Signaling 7 (C) • DSN ISDN User-to-Network Signaling (R) • Application (R) • Physical Layer (R) • Data Link Layer (R) • Data Link Connection (R) • Peer-to-Peer Procedures of Data-Link Layer (R) • Layer 3 DSN User-to-Network Signaling (R) • DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R) • Sequence of Messages for DSN Circuit-Switched Calls (R) • Message Functional Definition and Content (R) • General Message Format and Information Elements Coding (R) • Supplementary Services (C) • PCM-24 Digital Trunk Interface (R) • PCM-30 Digital Trunk Interface (Europe only) (R) • Interoperation of PCM-24 and PCM-30 (R) • Analog Trunk Interface (C) • Integrated Digital Loop Carrier (C) • Local Office Test Line (C) • Outside Plant Test Lines (C) • Test Incoming Trunks Tandem or Local State (C) • Manual Test of Trunks (R) • Trunk Group-Remove from Service (R) • Trunk Group-Restore to Service (R) • Carrier Group Alarm (R) • Software Carrier Group Alarm (C)	• UCR Section 2.3.2 • UCR Section 2.3.4.1 • UCR Section 2.3.4.1.1 • UCR Section 2.3.4.2 • UCR Section 2.3.4.2.1 • UCR Section 5.3.3.1.1 • UCR Section 5.3.3.1.2 • UCR Section 5.3.3.2.1 • UCR Section 5.3.3.2.2 • UCR Section 5.3.5 • UCR Section 5.3.6 • UCR Section 5.3.7 • UCR Section 5.3.8 • UCR Section 5.3.9 • UCR Section 5.3.10 • UCR Section 5.4.1 • UCR Section 5.4.2 • UCR Section 5.4.2.1 • UCR Section 5.4.3 • UCR Section 5.5 • UCR Section 5.6 • UCR Section 5.7.1 • UCR Section 5.7.1.1 • UCR Section 5.7.1.2 • UCR Section 5.7.1.3 • UCR Section 5.7.1.3.1 • UCR Section 5.7.1.3.2 • UCR Section 5.7.1.4 • UCR Section 5.7.1.4.2 • UCR Section 5.7.1.4.3 • UCR Section 5.7.1.4.4 • UCR Section 5.7.1.4.5 • UCR Section 5.7.1.4.6 • UCR Section 7.1 • UCR Section 7.2 • UCR Section 7.3 • UCR Section 7.4 • UCR Section 7.5 • UCR Section 2.5.1 • UCR Section 2.5.2 • UCR Section 2.5.3 • UCR Section 2.5.4.2 • UCR Section 2.5.5 • UCR Section 2.5.6 • UCR Section 2.5.7 • UCR Section 2.5.7.1

Table 2. SMEO Requirements (continued)

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 SS7 (ANSI T1.619a)	No	Voice	• MOS (R) • Secure calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
E1 SS7 (ITU-T Q.735.3)	No (Europe only)	Facsimile	• Analog: ITU-T T.4 (R)	• DISR
T1 CAS (MFR1)	No	Data	• Modem (VBD) (R) • 56 kbps switched data (R: PRI only) • 64 kbps switched data (R: PRI only) • NX56 synchronous BER (R: PRI only) • NX64 synchronous BER (R: PRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01C • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • CJCSI 6215.01C
T1 CAS (DTMF, DP)	Yes			
E1 CAS (MFR1)	No (Europe only)	VTC	• ITU-T H.320 (R: PRI only)	• FTR 1080B-2002
E1 CAS (DTMF, DP)	Yes (Europe only)			
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes			
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe Only)			
DSN Line Interfaces				
2-Wire Analog	Yes	Access	• Directory Number Identification (R) • PBX Line (C) • National ISDN 1/2 Basic Access (R) • Analog Line (R) • Basic Line Test Capabilities (R) • Advanced Line Test Capabilities (C) • Network Power Systems for External Interfaces (R) • Loop Start Line (R: 2-Wire Analog only) • Reverse Battery (R) • Alerting Signals and Tones (R) • S/T Reference Point (R)	• UCR Section 2.1.1 • UCR Section 2.3.1 • UCR Section 2.3.3 • UCR Section 2.3.5 • UCR Section 2.5.4.1.1 • UCR Section 2.5.4.1.2 • UCR Section 5.1 • UCR Section 5.2.1 • UCR Section 5.3.1 • UCR Section 5.5 • UCR Section 5.7.1.2.1
ISDN BRI NI 1/2 (ANSI T1.619a)	Yes			
2W Digital Proprietary	No	Voice	• MOS (R) • Secure Calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
VoIP	No	Data	• Modem (VBD) (R) • 56 kbps switched data (R) • 64 kbps switched data (R: BRI only) • NX56 synchronous BER (R: BRI only) • NX64 synchronous BER (R: BRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01C • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: BRI only)	• FTR 1080B-2002

Table 2. SMEO Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Common Features	Yes	• Individual Lines (R) • Selective call rejection (C) • Denied originating service (C) • Code restriction and diversion (R) • Call waiting (R) • Three-way calling (R) • Add-on transfer, conference calling, and call hold (C) • Call Transfer Individual – All calls (R) • Call Transfer - Internal Only (R) • Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R) • Call Transfer – Outside (R) • Call Transfer – Add-On Restricted Station (C) • Call Transfer – Attendant (C) • Call Hold (R) • Conference Calling – Six Way Station Controlled (C) • Call forwarding Variable (R) • Call Forward Busy Line (R) • Call Forwarding – Don't Answer – All Calls (R) • Selective Call Forwarding (C) • Call pick-up (C) • Address Translation (C) • Assured Dial Tone (R)	• UCR Section 2.1 • UCR Section 2.1.2 • UCR Section 2.1.3 • UCR Section 2.1.4 • UCR Section 2.1.5 • UCR Section 2.1.6 • UCR Section 2.1.7 • UCR Section 2.1.7.1 • UCR Section 2.1.7.2 • UCR Section 2.1.7.3 • UCR Section 2.1.7.4 • UCR Section 2.1.7.5 • UCR Section 2.1.7.6 • UCR Section 2.1.7.7 • UCR Section 2.1.7.8 • UCR Section 2.1.8.1 • UCR Section 2.1.8.2 • UCR Section 2.1.8.3 • UCR Section 2.1.8.4 • UCR Section 2.1.9 • UCR Section 2.7 • UCR Section 2.9
Attendant	No	• Attendant Features (C)	• UCR Section 2.2
Public Safety	Yes	• Basic Emergency Service (911) Caller (R) • Emergency Service (911) Public Safety Answering Point (C) • Enhanced Emergency Service (E911) (C) • Trace of terminating calls (R) • Outgoing call trace (R) • Tandem call trace (R) • Trace of a call in progress (R)	• UCR Section 2.4.1.1 • UCR Section 2.4.1.2 • UCR Section 2.4.1.3 • UCR Section 2.4.2 • UCR Section 2.4.3 • UCR Section 2.4.4 • UCR Section 2.4.5
Conferencing	Yes	• Preset Conferencing (C) • Meet-Me Conferencing (R) • Progressive Conferencing (C)	• UCR Section 2.6. • UCR Section 2.6.2 • UCR Section 2.6.3
Nailed-up Connections	No	• Nailed-Up Connection (C)	• UCR Section 2.8
DSN Hotline Services	Yes	• DSN Analog Hotline Service (R)	• UCR Section 2.12

Table 2. SMEO Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
MLPP	Yes	• MLPP Overview (R) • Preemption in the Network (R) • Network Facility with Lower Precedence Calls (R) • Cancel to / Cancel from (C) • Network Facility with Equal or Higher Precedence Calls (R) • MLPP Trunk Selection (R) • Hunt Sequence for Trunks (R) • ROUTINE Precedence Calls (R) • Precedence Calls Above ROUTINE Precedence (R) • Method 1 (R) • Method 2 (C) • MLPP Internetworking with other Networks (R) • Precedence Call Diversion (R) • Channel Associated Signaling (R) • Primary Rate Interface (R) • Common Channel Signaling Number 7 (C) • Analog Line MLPP (R) • ISDN MLPP Basic Rate Interface General Description (R) • Single B Channel, Single Appearance, Single DN (R) • Line Active with a Lower Precedence Call (R) • Line Active with a Equal or Higher Precedence Call (R) • Single B Channel, Multiple Appearances, Single DN (C) • Two B Channels, Multiple Appearances, Single DN (C) • Two B Channel, Two DN (Data Mode Only) (R) • ISDN Primary Rate Interface (R) • Precedence Call Waiting (R) • Call Forwarding (R) • Call Transfer (R) • Call Hold (R) • Three-Way Calling (R) • Call Pickup (C) • Conferencing (C) • Multiline Hunt Group (C) • Community of Interest (R) • MLPP Common Channel Signaling Number 7 (C) • CAS to CCS Trunk Network in a Mixed Media Network (C) • MLPP Interaction with EKTS features (C)	• UCR Section 3.1 • UCR Section 3.2 • UCR Section 3.2.1 • UCR Section 3.2.1.1 • UCR Section 3.2.2 • UCR Section 3.2.3 • UCR Section 3.2.3.1 • UCR Section 3.2.3.1.1 • UCR Section 3.2.3.1.2 • UCR Section 3.2.3.1.2.1 • UCR Section 3.2.3.1.2.2 • UCR Section 3.2.4 • UCR Section 3.3 • UCR Section 3.4.1 • UCR Section 3.4.2 • UCR Section 3.4.3 • UCR Section 3.5 • UCR Section 3.6.1 • UCR Section 3.6.2 • UCR Section 3.6.2.1 • UCR Section 3.6.2.2 • UCR Section 3.6.3 • UCR Section 3.6.4 • UCR Section 3.6.5 • UCR Section 3.7 • UCR Section 3.8.1 • UCR Section 3.8.2 • UCR Section 3.8.3 • UCR Section 3.8.4 • UCR Section 3.8.5 • UCR Section 3.8.6 • UCR Section 3.8.7 • UCR Section 3.8.8 • UCR Section 3.8.9 • UCR Section 3.9 • UCR Section 3.10 • UCR Section 3.11

Table 2. SMEO Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Call Processing	Yes	• Call Treatments (R) • Primary and Alternate Routing (R) • E&M Lead Signaling States (C) • 4-Wire Analog User Access Lines (C) • 2-Wire User Access Lines (R) • Termination of Analog Lines (R) • DSN Interswitch Trunk Call Processing (non-CCS/ISDN) (R) • DSN User Dialing (R) • Interswitch and Intraswitch Dialing (R) • Seven-Digit Dialing (R) • Ten-Digit Dialing (R) • Access Code (R) • Access Digit (R) • Precedence Digit (R) • Service Digit (R) • Route Code (R) • Area Code (R) • Switch Code (R) • Line Number (R) • Calling Name Delivery (C) • Calling Number Delivery (R) • Emergency Service 911 Conflict Resolution (R) • DSN Switch Outpulsing Digit Formats (C) • Standard Directory Number (R) • Standard Test Numbers (C) • Base Services – Abbreviated Numbers (R) • Digit Reception Requirements (R) • Digit Registration Capacity (R) • Screening (R)	• UCR Section 4.1 • UCR Section 4.2 • UCR Section 4.3.1 • UCR Section 4.3.2 • UCR Section 4.3.3 • UCR Section 4.3.4 • UCR Section 4.4 • UCR Section 4.5.1.1 • UCR Section 4.5.1.2 • UCR Section 4.5.1.2.1 • UCR Section 4.5.1.2.2 • UCR Section 4.5.1.3 • UCR Section 4.5.1.3.1 • UCR Section 4.5.1.3.2 • UCR Section 4.5.1.3.3 • UCR Section 4.5.1.4 • UCR Section 4.5.1.5 • UCR Section 4.5.1.6 • UCR Section 4.5.1.7 • UCR Section 4.5.1.8.1 • UCR Section 4.5.1.8.2 • UCR Section 4.5.1.9 • UCR Section 4.5.2 • UCR Section 4.5.3 • UCR Section 4.5.4 • UCR Section 4.5.5 • UCR Section 4.5.6 • UCR Section 4.5.7 • UCR Section 4.5.8
Network Management	Yes	• Interfaces (R) • Data Quality (R) • Traffic Measurements (R) • Reference Data (C) • Line Servicing (C) • Trunk Groups (C) • Call Processors (C) • Switch Services (C) • Special Studies (C) • Remote Switching Studies (C) • Features (C) • Common Channel Signaling Network Measurements (C) • ISDN Measurements (C) • Traffic Capacity (R) • Fault management (R) • Configuration management (R) • Call Detail Recording Data Retention (C) • Performance management (R) • Network Management controls (C) • Remote access (R)	• UCR Section 9.1 • UCR Section 9.2.1 • UCR Section 9.2.2.1.1 • UCR Section 9.2.2.1.2 • UCR Section 9.2.2.2 • UCR Section 9.2.2.3 • UCR Section 9.2.2.4 • UCR Section 9.2.2.5 • UCR Section 9.2.2.6 • UCR Section 9.2.2.7 • UCR Section 9.2.2.8 • UCR Section 9.2.3 • UCR Section 9.2.4 • UCR Section 9.2.5 • UCR Section 9.3 • UCR Section 9.4 • UCR Section 9.5.2 • UCR Section 9.6 • UCR Section 9.7 • UCR Section 9.8

Table 2. SMEO Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
ISDN Services	Yes	• BRI Access, Call Control and Signaling (R) • Uniform Interface Configuration for BRIs (R) • Electronic Key Telephone Systems (EKTS) (C) • PRI Access, Call Control and Signaling (R) • PRI Features (R) • Packet Data Features and Capabilities (C)	• UCR Section 10, Table 10-1 • UCR Section 10, Table 10-2 • UCR Section 10, Table 10-3 • UCR Section 10, Table 10-4 • UCR Section 10, Table 10-5 • UCR Section 10, Table 10-6
Synchronization	Yes	• External Timing Mode (C) • Line timing mode (R) • General (C) • Internal Stratum 4 (R) • Synchronization Performance Monitoring Criteria (C) • DS1 Traffic Interfaces (C) • DS0 Traffic Interconnects (C)	• UCR Section 11.1.1.1 • UCR Section 11.1.1.2 • UCR Section 11.1.2.1 • UCR Section 11.1.2.2 • UCR Section 11.2 • UCR Section 11.3 • UCR Section 11.4
Reliability (See note 1.)	Yes	• Reliability Requirements (R) • Backup Power (R) • Power Components (R) • UPS Requirements (R) • UPS Load Capacity (R) • Backup Power (Environmental) (R) • Alarms (R)	• UCR Section 12.1 • UCR Section 12.3 • UCR Section 12.3.1 • UCR Section 12.3.2 • UCR Section 12.3.2.1 • UCR Section 12.3.3 • UCR Section 12.3.4
Security	Yes	• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	• UCR Section 13
VoIP			
VoIP System	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with MOS of 4.0 or better (R) • ITU-T G.711 PCM CODEC (R) • MLPP • Security (R) • Network management (R) • System timing (R) • Latency ≤ 60 milliseconds (R) • IPv6 capable (R) • Service Class Tagging (R) • VoIP System Downtime (IP network 35 min/yr Subscriber 12 min/yr) (R)	• UCR App. 3, para. A3.2.1 • UCR App. 3, para. A3.2.2 • UCR App. 3, para. A3.2.3 • UCR App. 3, para. A3.2.4 • UCR App. 3, para. A3.2.5 • UCR App. 3, para. A3.2.6 • UCR App. 3, para. A3.2.7 • UCR App. 3, para. A3.2.8 • UCR App. 3, para. A3.2.9 • UCR App. 3, para. A3.2.10

Table 2. SMEO Requirements (continued)

Network Gateways				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
PSTN (See note 2.)	Yes	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) • Ground Start Line (R) • Immediate Start (C) • Delay Dial (C)	• CJCSI 6215.01C • CJCSI 6215.01C • CJCSI 6215.01C • UCR Section 5.2.2 • UCR Section 5.3.2 • UCR Section 5.3.4

NOTES:
1 Backup power, power components, UPS requirements, UPS load capacity and alarms are non-testable requirements. It is the responsibility of the respective base/post/camp/station communication agency to provide this with the SUT when installed.
2 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.
3 Data and VTC services are not provided via the DSN to tactical interface.

LEGEND:

2W	2-Wire	FTR 1080B-2002	Video Teleconferencing Services	PCM	Pulse Code Modulation
ANSI	American National Standards Institute	G.711	Standard for PCM of Voice Frequencies	PCM-24	Pulse Code Modulation - 24 Channels
App	Appendix	GR	Generic Requirement (Telcordia)	PCM-30	Pulse Code Modulation - 30 Channels
BER	Bit Error Ratio	GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security	PRI	Primary Rate Interface
BRI	Basic Rate Interface	H.320	Standard for Narrowband VTC	PSTN	Public Switched Telephone Network
C	Conditional	IP	Internet Protocol	Q.735.3	SS7 Signaling Standard for E1 MLPP
CAS	Channel Associated Signaling	IPv6	Internet Protocol version 6	Q.955.3	ISDN Signaling Standard for E1 MLPP
CCS	Common Channel Signaling	ISDN	Integrated Services Digital Network	R	Required
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	IT	Information Technology	SMEO	Small End Office
DIACAP	DoD Information Assurance Certification and Accreditation Process	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	SS7	Signaling System 7
DISR	DoD IT Standards Registry	kbps	kilobits per second	STE	Secure Terminal Equipment
DoD	Department of Defense	Mbps	Megabits per second	STIGs	Security Technical Implementation Guides
DoDI	Department of Defense Instruction	MFR1	Multi-Frequency	STU-III	Secure Telephone Unit – 3rd Generation
DP	Dial Pulse	min	Recommendation 1 minute	S/T	ISDN BRI 4-wire interface
DN	Directory Number	MLPP	Multi-Level Precedence and Preemption	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DS0	Digital Signal Level 0 (64 kbps)	MOS	Mean Opinion Score	T.4	Standardization of Group 3 facsimile terminals for document transmission
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	NI 1/2	National ISDN Standard 1 or 2	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DSN	Defense Switched Network	NX56	Data format restricted to multiples of 56 kbps	TIA	Telecommunications Industry Association
DTMF	Dual Tone Multi-Frequency	NX64	Data format restricted to multiples of 64 kbps	UCR	Unified Capabilities Requirements
E&M	Ear and Mouth	para	paragraph	UPS	Uninterruptible Power Supply
E1	European Basic Multiplex Rate (2.048 Mbps)	PBX	Private Branch Exchange	VBD	Variable bit data
EKTS	Electronic Key Telephone System			VoIP	Voice over Internet Protocol
FTR	Federal Telecommunications Recommendation			VTC	Video Teleconferencing
				yr	year

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and

references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet), or http://199.208.204.125 (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi.

6. The JITC point of contact is Mr. Edward Mellon, DSN 879-5269, commercial (520) 538-5159, FAX DSN 879-4347, or e-mail to edward.mellon@disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The tracking number for the SUT is 0819002.

FOR THE COMMANDER:

2 Enclosures a/s

for RICHARD A. MEADOR
Chief
Battlespace Communications Portfolio

Distribution (electronic mail):
Joint Staff J-6
Joint Interoperability Test Command, Liaison, TE3/JT1
Office of Chief of Naval Operations, CNO N6F2
Headquarters U.S. Air Force, Office of Warfighting Integration & CIO, AF/XCIN (A6N)
Department of the Army, Office of the Secretary of the Army, DA-OSA CIO/G-6 ASA (ALT), SAIS-IOQ
U.S. Marine Corps MARCORSYSCOM, SIAT, MJI Division I
DOT&E, Net-Centric Systems and Naval Warfare
U.S. Coast Guard, CG-64
Defense Intelligence Agency
National Security Agency, DT
Defense Information Systems Agency, TEMC
Office of Assistant Secretary of Defense (NII)/DOD CIO
U.S. Joint Forces Command, Net-Centric Integration, Communication, and Capabilities Division, J68
Defense Information Systems Agency, GS23

ADDITIONAL REFERENCES

(c) Joint Interoperability Test Command, "Information Assurance (IA) Assessment of PacStar 6800 Small End Office (SMEO) IQ-Core version (v)3.0 (Tracking Number 0819002)," 11 August 2009

(d) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, "Policy for Department of Defense Voice Services with Real Time Services (RTS)," 9 November 2007

(e) Defense Information Systems Agency, "Department of Defense Networks Unified Capabilities Requirements," 21 December 2007

(f) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006

CERTIFICATION TESTING SUMMARY

1. SYSTEM TITLE. PacStar 6800 Enterprise Unified Capabilities (UC) Exchange with software version IQ-Core 3.0; hereinafter referred to as the System Under Test (SUT).

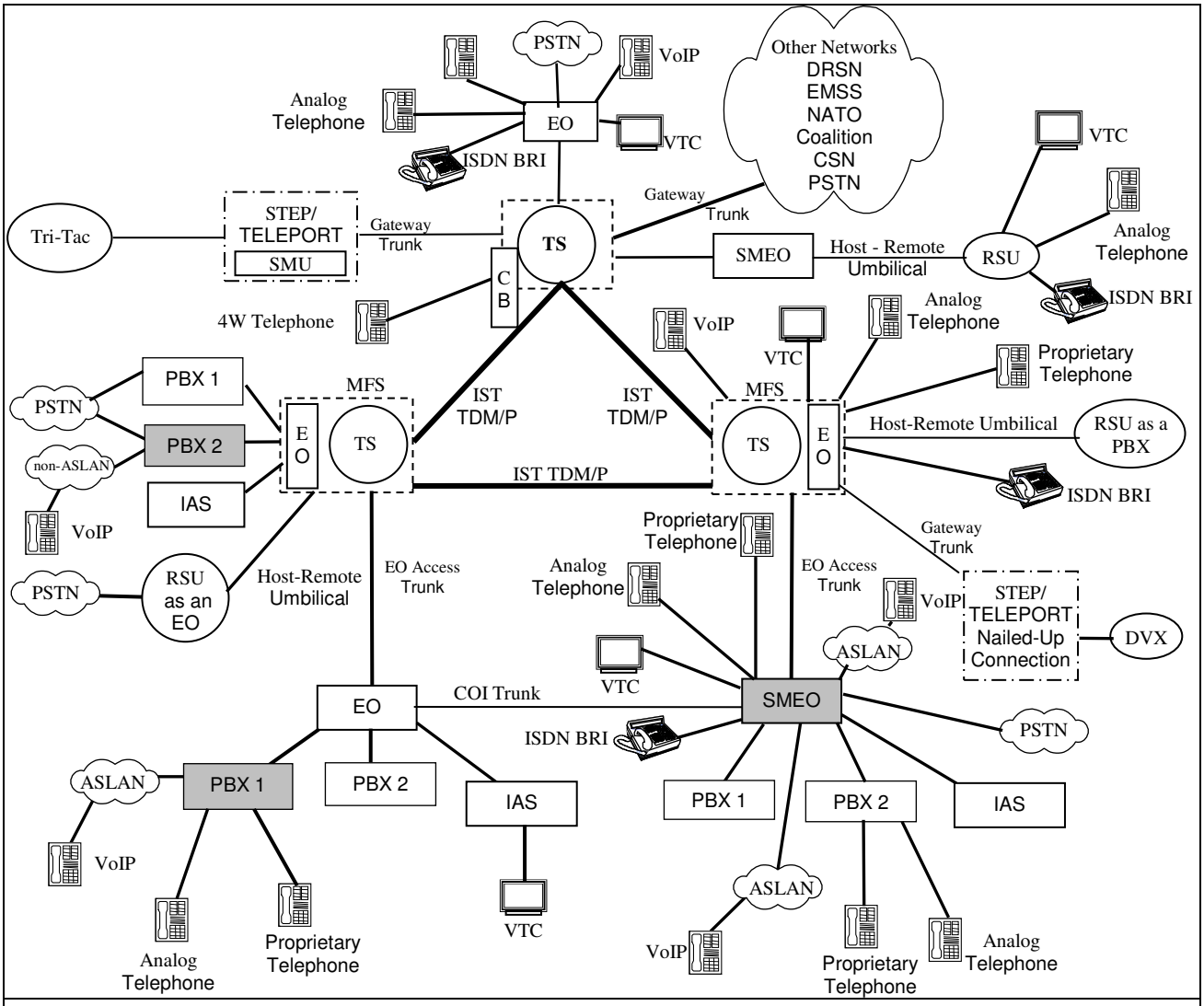
2. PROPONENT. United Joint Communications Support (JCSE), Systems Acquisition Directorate.

3. PROGRAM MANAGER. John Dunn YA-02 JCSE J5, Chief of Technology Officer (CTO), 8532 Marina Bay Drive MacDill AFB FL, 33621-5504, e-mail: John.Dunn@JCSE.MIL.

4. TESTER. Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.

5. SYSTEM UNDER TEST DESCRIPTION. The SUT is a Small End Office (SMEO). The SUT is comprised of two major enclaves, a REDCOM enclave and a Cisco CallManager enclave. The SUT provides both Time Division Multiplexing (TDM) digital telecommunications switching and Voice over Internet Protocol (VoIP). The SUT Cisco CallManager enclave consists of Cisco Media Convergence Server (MCS)7825, MCS7835, and/or MCS7845 series servers running the Cisco Unified CallManager software, Cisco 3825 and/or 3845 gateways, and Internet Protocol (IP) telephones. Two Cisco Catalyst 3750 series switches and/or a Cisco NME-16ES-1G Switch Module provides internal LAN switching among the SUT components. The two enclaves are connected via Digital Transmission Link Level 1 (T1) Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI). The SUT supports the following line interfaces: analog, Skinny Client Control Protocol (SCCP)-based VoIP, International Telecommunication Union – Telecommunication Standardization Sector (ITU-T) V.150.1 Modem over Internet Protocol (MoIP) protocol, and digital ISDN Basic Rate Interface (BRI). The SUT supports the following trunk interfaces: T1 ISDN PRI, European Basic Multiplex Rate (E1) PRI, T1 Channel Associated Signaling (CAS), E1 CAS. PacStar IQ-Core software monitors and manages the entire 6800 system. The SUT offers the following features: scalable, distributed platform for growth of up to a maximum of 7,900 TDM and 30,000 IP subscribers, modular client/server architecture for flexibility, scalability, and a redundant call processing core for extra reliability in mission-critical enterprises. The SUT is certified for VoIP with any certified Assured Services Local Area Networks (ASLANs) posted on the UC Approved Products List (APL).

6. OPERATIONAL ARCHITECTURE. The DSN architecture is a two-level network hierarchy consisting of DSN backbone switches and Service/Agency installation switches. Joint Staff policy and subscriber mission requirements determine which type of switch can be used at a particular location. The DSN architecture, therefore, consists of several categories of switches including SMEOs. The Unified Capabilities Requirements (UCR) operational DSN Architecture is depicted in Figure 2-1. This architecture depicts the relationship of Military Department SMEOs to the other DSN switch types.



LEGEND:

4W	4-Wire
ASLAN	Assured Services Local Area Network
BRI	Basic Rate Interface
CB	Channel Bank
COI	Community of Interest
CSN	Canadian Switch Network
DRSN	Defense Red Switch Network
DSN	Defense Switched Network
DVX	Deployable Voice Exchange
EMSS	Enhanced Mobile Satellite System
EO	End Office
IAS	Integrated Access Switch
ISDN	Integrated Services Digital Network
IST	Interswitch Trunk
MFS	Multifunction Switch
NATO	North Atlantic Treaty Organization
PBX	Private Branch Exchange
PBX 1	Private Branch Exchange 1
PBX 2	Private Branch Exchange 2
PSTN	Public Switched Telephone Network
RSU	Remote Switching Unit
SMEO	Small End Office
SMU	Switched Multiplex Unit
STEP	Standardized Tactical Entry Point
TDM/P	Time Division Multiplex/Packetized
Tri-Tac	Tri-Service Tactical Communications Program
TS	Tandem Switch
VoIP	Voice over Internet Protocol
VTC	Video Teleconferencing

Figure 2-1. DSN Architecture

7. REQUIRED SYSTEM INTERFACES. Requirements specific to SMEOs are listed in Table 2-1. These requirements are derived from:

a. DSN services for Network and Applications specified in Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, "Policy for Department of Defense Voice Services for Real Time Services (RTS)."

b. UCR interface and signaling requirements for trunks/lines verified through JITC testing and/or vendor submission of Letters of Compliance (LoC).

c. UCR SMEO Capability Requirements (CRs) and Feature Requirements (FRs) verified through JITC testing and/or vendor submission of LoC.

Table 2-1. SMEO Requirements

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 SS7 (ANSI T1.619a) E1 SS7 (ITU-T Q.735.3) T1 CAS (MFR1) T1 CAS (DTMF, DP) E1 CAS (DTMF, DP) E1 CAS (MFR1) T1 ISDN PRI NI 1/2 (ANSI T1.619a) E1 ISDN PRI (ITU-T Q.955.3)	No No (Europe only) No Yes Yes (Europe only) No (Europe only) Yes No (Europe Only)	Trunking	• Direct Inward Dialing (C) • National ISDN 1/2 Primary Access (R) • ISDN ANSI MLPP Service Capability (R) • ITU-T ISDN Primary Access (Europe only) (C) • ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (Europe only) (C) • Normal Wink Start Operations (R) • Glare Operation (R) • Abnormal Wink Start (R) • Glare Resolution (R) • Call for Service Timing (R) • Guard Timing (R) • Satellite Timing (R) • Disconnect Control (R) • Reselect and Retrial (R) • Off-Hook Supervision Transition (R) • Dial-Pulse Signals (R) • DTMF Signaling (R) • Standard Digit Format for Precedence (C) • MFR1 2/6 Signaling (C) • Alerting Signals and Tones (R) • Common Channel Signaling 7 (C) • DSN ISDN User-to-Network Signaling (R) • Application (R) • Physical Layer (R) • Data Link Layer (R) • Data Link Connection (R) • Peer-to-Peer Procedures of Data-Link Layer (R) • Layer 3 DSN User-to-Network Signaling (R) • DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R) • Sequence of Messages for DSN Circuit-Switched Calls (R) • Message Functional Definition and Content (R) • General Message Format and Information Elements Coding (R) • Supplementary Services (C) • PCM-24 Digital Trunk Interface (R) • PCM-30 Digital Trunk Interface (Europe only) (R) • Interoperation of PCM-24 and PCM-30 (R)	• UCR Section 2.3.2 • UCR Section 2.3.4.1 • UCR Section 2.3.4.1.1 • UCR Section 2.3.4.2 • UCR Section 2.3.4.2.1 • UCR Section 5.3.3.1.1 • UCR Section 5.3.3.1.2 • UCR Section 5.3.3.2.1 • UCR Section 5.3.3.2.2 • UCR Section 5.3.5 • UCR Section 5.3.6 • UCR Section 5.3.7 • UCR Section 5.3.8 • UCR Section 5.3.9 • UCR Section 5.3.10 • UCR Section 5.4.1 • UCR Section 5.4.2 • UCR Section 5.4.2.1 • UCR Section 5.4.3 • UCR Section 5.5 • UCR Section 5.6 • UCR Section 5.7.1 • UCR Section 5.7.1.1 • UCR Section 5.7.1.2 • UCR Section 5.7.1.3 • UCR Section 5.7.1.3.1 • UCR Section 5.7.1.3.2 • UCR Section 5.7.1.4 • UCR Section 5.7.1.4.2 • UCR Section 5.7.1.4.3 • UCR Section 5.7.1.4.4 • UCR Section 5.7.1.4.5 • UCR Section 5.7.1.4.6 • UCR Section 7.1 • UCR Section 7.2 • UCR Section 7.3

Table 2-1. SMEO Requirements (continued)

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 SS7 (ANSI T1.619a) E1 SS7 (ITU-T Q.735.3) T1 CAS (MFR1) T1 CAS (DTMF, DP) E1 CAS (MFR1) E1 CAS (DTMF, DP) T1 ISDN PRI NI 1/2 (ANSI T1.619a) E1 ISDN PRI (ITU-T Q.955.3)	No No (Europe only) No Yes No (Europe only) Yes (Europe only) Yes No (Europe Only)	Trunking continued	• Analog Trunk Interface (C) • Integrated Digital Loop Carrier (C) • Local Office Test Line (C) • Outside Plant Test Lines (C) • Test Incoming Trunks Tandem or Local State (C) • Manual Test of Trunks (R) • Trunk Group-Remove from Service (R) • Trunk Group-Restore to Service (R) • Carrier Group Alarm (R) • Software Carrier Group Alarm (C)	• UCR Section 7.4 • UCR Section 7.5 • UCR Section 2.5.1 • UCR Section 2.5.2 • UCR Section 2.5.3 • UCR Section 2.5.4.2 • UCR Section 2.5.5 • UCR Section 2.5.6 • UCR Section 2.5.7 • UCR Section 2.5.7.1
		Voice	• MOS (R) • Secure calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R) • 56 kbps switched data (R: PRI only) • 64 kbps switched data (R: PRI only) • NX56 synchronous BER (R: PRI only) • NX64 synchronous BER (R: PRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01C • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: PRI only)	• FTR 1080B-2002
DSN Line Interfaces				
2-Wire Analog ISDN BRI NI 1/2 (ANSI T1.619a) 2W Digital Proprietary VoIP	Yes Yes No No	Access	• Directory Number Identification (R) • PBX Line (C) • National ISDN 1/2 Basic Access (R) • Analog Line (R) • Basic Line Test Capabilities (R) • Advanced Line Test Capabilities (C) • Network Power Systems for External Interfaces (R) • Loop Start Line (R: 2-Wire Analog only) • Reverse Battery (R) • Alerting Signals and Tones (R) • S/T Reference Point (R)	• UCR Section 2.1.1 • UCR Section 2.3.1 • UCR Section 2.3.3 • UCR Section 2.3.5 • UCR Section 2.5.4.1.1 • UCR Section 2.5.4.1.2 • UCR Section 5.1 • UCR Section 5.2.1 • UCR Section 5.3.1 • UCR Section 5.5 • UCR Section 5.7.1.2.1
		Voice	• MOS (R) • Secure Calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R) • 56 kbps switched data (R) • 64 kbps switched data (R: BRI only) • NX56 synchronous BER (R: BRI only) • NX64 synchronous BER (R: BRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01C • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: BRI only)	• FTR 1080B-2002

Table 2-1. SMEO Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Common Features	Yes	• Individual Lines (R) • Selective call rejection (C) • Denied originating service (C) • Code restriction and diversion (R) • Call waiting (R) • Three-way calling (R) • Add-on transfer, conference calling, and call hold (C) • Call Transfer Individual – All calls (R) • Call Transfer - Internal Only (R) • Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R) • Call Transfer – Outside (R) • Call Transfer – Add-On Restricted Station (C) • Call Transfer – Attendant (C) • Call Hold (R) • Conference Calling – Six Way Station Controlled (C) • Call forwarding Variable (R) • Call Forward Busy Line (R) • Call Forwarding – Don't Answer – All Calls (R) • Selective Call Forwarding (C) • Call pick-up (C) • Address Translation (C) • Assured Dial Tone (R)	• UCR Section 2.1 • UCR Section 2.1.2 • UCR Section 2.1.3 • UCR Section 2.1.4 • UCR Section 2.1.5 • UCR Section 2.1.6 • UCR Section 2.1.7 • UCR Section 2.1.7.1 • UCR Section 2.1.7.2 • UCR Section 2.1.7.3 • UCR Section 2.1.7.4 • UCR Section 2.1.7.5 • UCR Section 2.1.7.6 • UCR Section 2.1.7.7 • UCR Section 2.1.7.8 • UCR Section 2.1.8.1 • UCR Section 2.1.8.2 • UCR Section 2.1.8.3 • UCR Section 2.1.8.4 • UCR Section 2.1.9 • UCR Section 2.7 • UCR Section 2.9
Attendant	No	• Attendant Features (C)	• UCR Section 2.2
Public Safety	Yes	• Basic Emergency Service (911) Caller (R) • Emergency Service (911) Public Safety Answering Point (C) • Enhanced Emergency Service (E911) (C) • Trace of terminating calls (R) • Outgoing call trace (R) • Tandem call trace (R) • Trace of a call in progress (R)	• UCR Section 2.4.1.1 • UCR Section 2.4.1.2 • UCR Section 2.4.1.3 • UCR Section 2.4.2 • UCR Section 2.4.3 • UCR Section 2.4.4 • UCR Section 2.4.5
Conferencing	Yes	• Preset Conferencing (C) • Meet-Me Conferencing (R) • Progressive Conferencing (C)	• UCR Section 2.6. • UCR Section 2.6.2 • UCR Section 2.6.3
Nailed-up Connections	No	• Nailed-Up Connection (C)	• UCR Section 2.8
DSN Hotline Services	Yes	• DSN Analog Hotline Service (R)	• UCR Section 2.12

Table 2-1. SMEO Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
MLPP	Yes	• MLPP Overview (R) • Preemption in the Network (R) • Network Facility with Lower Precedence Calls (R) • Cancel to / Cancel from (C) • Network Facility with Equal or Higher Precedence Calls (R) • MLPP Trunk Selection (R) • Hunt Sequence for Trunks (R) • ROUTINE Precedence Calls (R) • Precedence Calls Above ROUTINE Precedence (R) • Method 1 (R) • Method 2 (C) • MLPP Internetworking with other Networks (R) • Precedence Call Diversion (R) • Channel Associated Signaling (R) • Primary Rate Interface (R) • Common Channel Signaling Number 7 (C) • Analog Line MLPP (R) • ISDN MLPP Basic Rate Interface General Description (R) • Single B Channel, Single Appearance, Single DN (R) • Line Active with a Lower Precedence Call (R) • Line Active with a Equal or Higher Precedence Call (R) • Single B Channel, Multiple Appearances, Single DN (C) • Two B Channels, Multiple Appearances, Single DN (C) • Two B Channel, Two DN (Data Mode Only) (R) • ISDN Primary Rate Interface (R) • Precedence Call Waiting (R) • Call Forwarding (R) • Call Transfer (R) • Call Hold (R) • Three-Way Calling (R) • Call Pickup (C) • Conferencing (C) • Multiline Hunt Group (C) • Community of Interest (R) • MLPP Common Channel Signaling Number 7 (C) • CAS to CCS Trunk Network in a Mixed Media Network (C) • MLPP Interaction with EKTS features (C)	• UCR Section 3.1 • UCR Section 3.2 • UCR Section 3.2.1 • UCR Section 3.2.1.1 • UCR Section 3.2.2 • UCR Section 3.2.3 • UCR Section 3.2.3.1 • UCR Section 3.2.3.1.1 • UCR Section 3.2.3.1.2 • UCR Section 3.2.3.1.2.1 • UCR Section 3.2.3.1.2.2 • UCR Section 3.2.4 • UCR Section 3.3 • UCR Section 3.4.1 • UCR Section 3.4.2 • UCR Section 3.4.3 • UCR Section 3.5 • UCR Section 3.6.1 • UCR Section 3.6.2 • UCR Section 3.6.2.1 • UCR Section 3.6.2.2 • UCR Section 3.6.3 • UCR Section 3.6.4 • UCR Section 3.6.5 • UCR Section 3.7 • UCR Section 3.8.1 • UCR Section 3.8.2 • UCR Section 3.8.3 • UCR Section 3.8.4 • UCR Section 3.8.5 • UCR Section 3.8.6 • UCR Section 3.8.7 • UCR Section 3.8.8 • UCR Section 3.8.9 • UCR Section 3.9 • UCR Section 3.10 • UCR Section 3.11

Table 2-1. SMEO Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Call Processing	Yes	• Call Treatments (R) • Primary and Alternate Routing (R) • E&M Lead Signaling States (C) • 4-Wire Analog User Access Lines (C) • 2-Wire User Access Lines (R) • Termination of Analog Lines (R) • DSN Interswitch Trunk Call Processing (non-CCS/ISDN) (R) • DSN User Dialing (R) • Interswitch and Intraswitch Dialing (R) • Seven-Digit Dialing (R) • Ten-Digit Dialing (R) • Access Code (R) • Access Digit (R) • Precedence Digit (R) • Service Digit (R) • Route Code (R) • Area Code (R) • Switch Code (R) • Line Number (R) • Calling Name Delivery (C) • Calling Number Delivery (R) • Emergency Service 911 Conflict Resolution (R) • DSN Switch Outpulsing Digit Formats (C) • Standard Directory Number (R) • Standard Test Numbers (C) • Base Services – Abbreviated Numbers (R) • Digit Reception Requirements (R) • Digit Registration Capacity (R) • Screening (R)	• UCR Section 4.1 • UCR Section 4.2 • UCR Section 4.3.1 • UCR Section 4.3.2 • UCR Section 4.3.3 • UCR Section 4.3.4 • UCR Section 4.4 • UCR Section 4.5.1.1 • UCR Section 4.5.1.2 • UCR Section 4.5.1.2.1 • UCR Section 4.5.1.2.2 • UCR Section 4.5.1.3 • UCR Section 4.5.1.3.1 • UCR Section 4.5.1.3.2 • UCR Section 4.5.1.3.3 • UCR Section 4.5.1.4 • UCR Section 4.5.1.5 • UCR Section 4.5.1.6 • UCR Section 4.5.1.7 • UCR Section 4.5.1.8.1 • UCR Section 4.5.1.8.2 • UCR Section 4.5.1.9 • UCR Section 4.5.2 • UCR Section 4.5.3 • UCR Section 4.5.4 • UCR Section 4.5.5 • UCR Section 4.5.6 • UCR Section 4.5.7 • UCR Section 4.5.8
Network Management	Yes	• Interfaces (R) • Data Quality (R) • Traffic Measurements (R) • Reference Data (C) • Line Servicing (C) • Trunk Groups (C) • Call Processors (C) • Switch Services (C) • Special Studies (C) • Remote Switching Studies (C) • Features (C) • Common Channel Signaling Network Measurements (C) • ISDN Measurements (C) • Traffic Capacity (R) • Fault management (R) • Configuration management (R) • Call Detail Recording Data Retention (C) • Performance management (R) • Network Management controls (C) • Remote access (R)	• UCR Section 9.1 • UCR Section 9.2.1 • UCR Section 9.2.2.1.1 • UCR Section 9.2.2.1.2 • UCR Section 9.2.2.2 • UCR Section 9.2.2.3 • UCR Section 9.2.2.4 • UCR Section 9.2.2.5 • UCR Section 9.2.2.6 • UCR Section 9.2.2.7 • UCR Section 9.2.2.8 • UCR Section 9.2.3 • UCR Section 9.2.4 • UCR Section 9.2.5 • UCR Section 9.3 • UCR Section 9.4 • UCR Section 9.5.2 • UCR Section 9.6 • UCR Section 9.7 • UCR Section 9.8

Table 2-1. SMEO Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
ISDN Services	Yes	• BRI Access, Call Control and Signaling (R) • Uniform Interface Configuration for BRIs (R) • Electronic Key Telephone Systems (EKTS) (C) • PRI Access, Call Control and Signaling (R) • PRI Features (R) • Packet Data Features and Capabilities (C)	• UCR Section 10, Table 10-1 • UCR Section 10, Table 10-2 • UCR Section 10, Table 10-3 • UCR Section 10, Table 10-4 • UCR Section 10, Table 10-5 • UCR Section 10, Table 10-6
Synchronization	Yes	• External Timing Mode (C) • Line timing mode (R) • General (C) • Internal Stratum 4 (R) • Synchronization Performance Monitoring Criteria (C) • DS1 Traffic Interfaces (C) • DS0 Traffic Interconnects (C)	• UCR Section 11.1.1.1 • UCR Section 11.1.1.2 • UCR Section 11.1.2.1 • UCR Section 11.1.2.2 • UCR Section 11.2 • UCR Section 11.3 • UCR Section 11.4
Reliability (See note 1.)	Yes	• Reliability Requirements (R) • Backup Power (R) • Power Components (R) • UPS Requirements (R) • UPS Load Capacity (R) • Backup Power (Environmental) (R) • Alarms (R)	• UCR Section 12.1 • UCR Section 12.3 • UCR Section 12.3.1 • UCR Section 12.3.2 • UCR Section 12.3.2.1 • UCR Section 12.3.3 • UCR Section 12.3.4
Security	Yes	• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	• UCR Section 13
VoIP			
VoIP System	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with MOS of 4.0 or better (R) • ITU-T G.711 PCM CODEC (R) • MLPP • Security (R) • Network management (R) • System timing (R) • Latency ≤ 60 milliseconds (R) • IPv6 capable (R) • Service Class Tagging (R) • VoIP System Downtime (IP network 35 min/yr Subscriber 12 min/yr) (R)	• UCR App. 3, para. A3.2.1 • UCR App. 3, para. A3.2.2 • UCR App. 3, para. A3.2.3 • UCR App. 3, para. A3.2.4 • UCR App. 3, para. A3.2.5 • UCR App. 3, para. A3.2.6 • UCR App. 3, para. A3.2.7 • UCR App. 3, para. A3.2.8 • UCR App. 3, para. A3.2.9 • UCR App. 3, para. A3.2.10

Table 2-1. SMEO Requirements (continued)

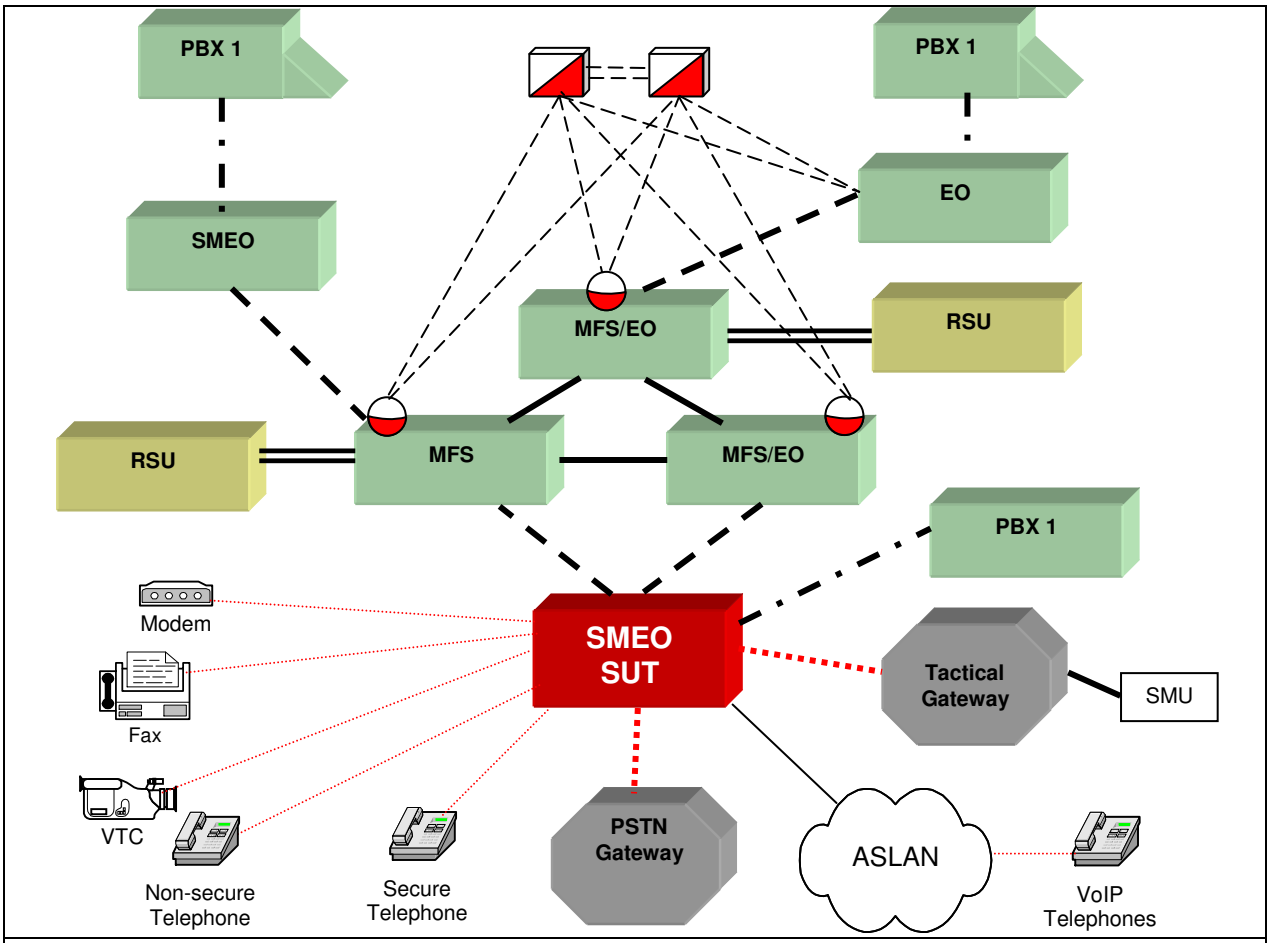
Network Gateways				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
PSTN (See note 2.)	Yes	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) • Ground Start Line (R) • Immediate Start (C) • Delay Dial (C)	• CJCSI 6215.01C • CJCSI 6215.01C • CJCSI 6215.01C • UCR Section 5.2.2 • UCR Section 5.3.2 • UCR Section 5.3.4

NOTES:
1 Backup power, power components, UPS requirements, UPS load capacity and alarms are non-testable requirements. It is the responsibility of the respective base/post/camp/station communication agency to provide this with the SUT when installed.
2 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.

LEGEND:

2W	2-Wire	FTR 1080B-2002	Video Teleconferencing Services	PBX	Private Branch Exchange
ANSI	American National Standards Institute	G.711	Standard for PCM of Voice Frequencies	PCM	Pulse Code Modulation
App	Appendix	GR	Generic Requirement (Telcordia)	PCM-24	Pulse Code Modulation - 24 Channels
BER	Bit Error Ratio	GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security	PCM-30	Pulse Code Modulation - 30 Channels
BRI	Basic Rate Interface	H.320	Standard for Narrowband VTC	PRI	Primary Rate Interface
C	Conditional	IP	Internet Protocol	PSTN	Public Switched Telephone Network
CAS	Channel Associated Signaling	IPv6	Internet Protocol version 6	Q.735.3	SS7 Signaling Standard for E1 MLPP
CCS	Common Channel Signaling	ISDN	Integrated Services Digital Network	Q.955.3	ISDN Signaling Standard for E1 MLPP
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	IT	Information Technology	R	Required
DIACAP	DoD Information Assurance Certification and Accreditation Process	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	SMEO	Small End Office
DISR	DoD IT Standards Registry	kbps	kilobits per second	SS7	Signaling System 7
DoD	Department of Defense	Mbps	Megabits per second	STE	Secure Terminal Equipment
DoDI	Department of Defense Instruction	MFR1	Multi-Frequency Recommendation 1	STIGs	Security Technical Implementation Guides
DP	Dial Pulse	min	minute	STU-III	Secure Telephone Unit – 3rd Generation
DN	Directory Number	MLPP	Multi-Level Precedence and Preemption	S/T	ISDN BRI 4-wire interface
DS0	Digital Signal Level 0 (64 kbps)	MOS	Mean Opinion Score	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	NI 1/2	National ISDN Standard 1 or 2	T.4	Standardization of Group 3 facsimile terminals for document transmission
DSN	Defense Switched Network	NX56	Data format restricted to multiples of 56 kbps	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DTMF	Dual Tone Multi-Frequency	NX64	Data format restricted to multiples of 64 kbps	TIA	Telecommunications Industry Association
E&M	Ear and Mouth	para	paragraph	UCR	Unified Capabilities Requirements
E1	European Basic Multiplex Rate (2.048 Mbps)			UPS	Uninterruptible Power Supply
EKTS	Electronic Key Telephone System			VBD	Variable bit data
FTR	Federal Telecommunications Recommendation			VoIP	Voice over Internet Protocol
				VTC	Video Teleconferencing
				yr	year

8. TEST NETWORK DESCRIPTION. The SUT was tested at JITC's Global Information Grid Network Test Facility in a manner and configuration similar to that of the DSN operational environment. Testing of the system's required functions and features was conducted using the test configuration depicted in Figure 2-2. The SUT was tested as the end-point in relation to the other switches. Figure 2-3 depicts the test configuration.



LEGEND:

A-Link	Access Link (SS7)
ASLAN	Assured Services Voice Application Local Area Network
B-Link	Bridge Link (SS7)
BRI	Basic Rate Interface
C-Link	Cross Link (SS7)
CAS	Channel Associated Signaling
DRSN	Defense Red Switch Network
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
EO	End Office
FAX	facsimile
ISDN	Integrated Services Digital Network
Mbps	Megabits per second
MFS	Multifunction Switch
PBX	Private Branch Exchange
PBX 1	Private Branch Exchange Type 1
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RSU	Remote Switching Unit
SMEO	Small End Office
SMU	Switch Multiplexer Unit
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TCP/IP	Transmission Control Protocol/Internet Protocol
VoIP	Voice over Internet Protocol
VTC	Video Teleconferencing

SS7 Service Switching Point (SSP)

SS7 Signal Transfer Point (STP)

DSN Gateway Trunk

DSN Interswitch Trunk (T1/E1 SS7, T1/E1 CAS, T1/E1 ISDN PRI)

DSN Line (2-Wire Analog, ISDN BRI, Digital Proprietary)

SS7 Links (A-Link, B-Link, or C-Link)

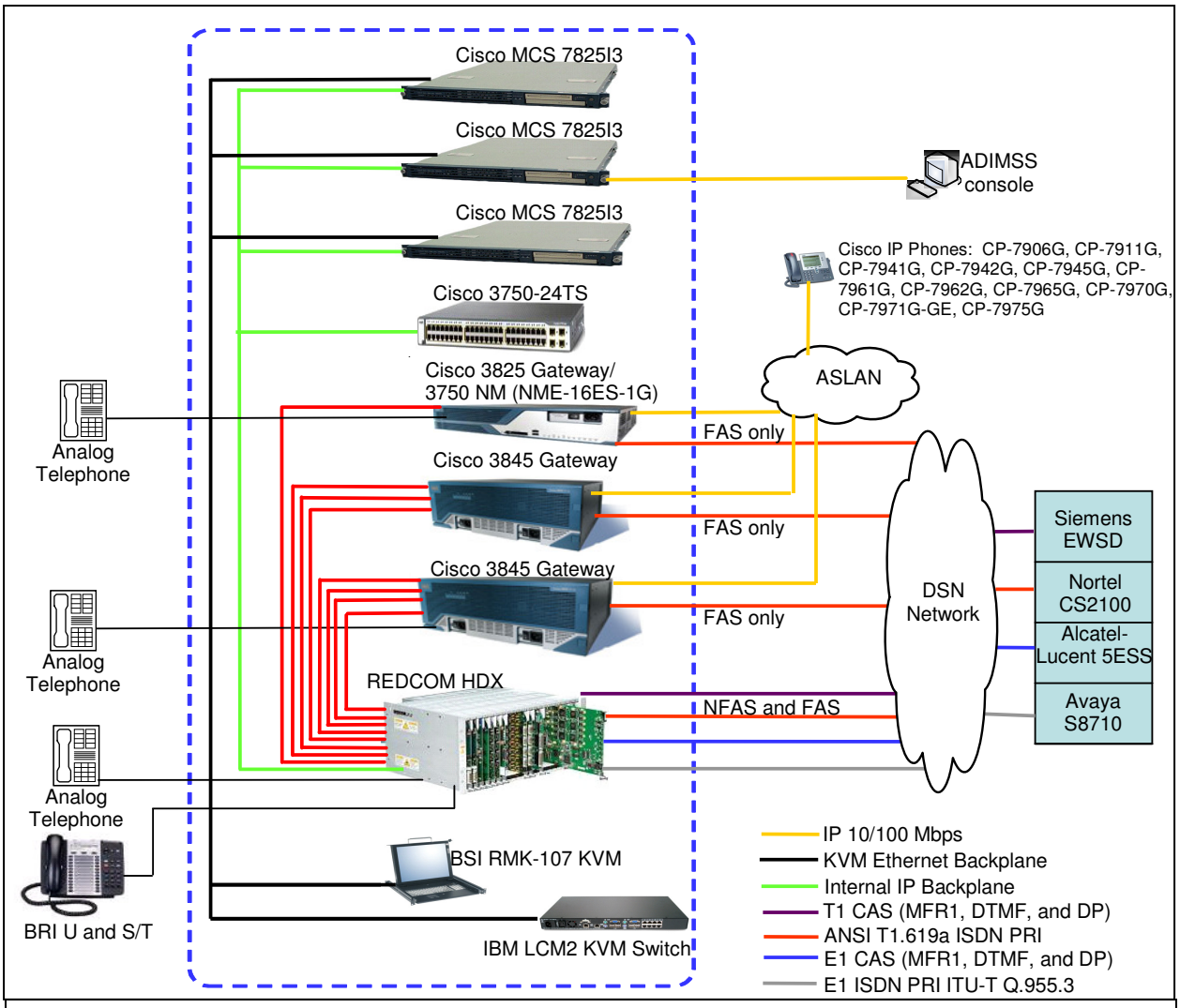
TCP/IP

DSN End Office Access Trunk (T1/E1 SS7, T1/E1 CAS, T1/E1 ISDN PRI)

DSN PBX Access Trunk (T1/E1 SS7, T1/E1 CAS, T1/E1 ISDN PRI)

RSU-Host Umbilical Link

Figure 2-2. Test Configuration



LEGEND:

5ESS	Class 5 Electronic Switching System
ADIMSS	Advanced DSN Integrated Management Support System
ANSI	American National Standards Institute
ASLAN	Assured Services Local Area Network
BRI	Basic Rate Interface
BSI	Broadax Systems Incorporated
CAS	Channel Associated Signaling
CS	Communication Server
DP	Dial Pulse
DSN	Defense Switched Network
DTMF	Dual Tone Multi-Frequency
E1	European Basic Multiplex Rate (2.048 Mbps)
EWSD	Elektronisches Wählsystem Digital
FAS	Facility Associated Signaling
HDX	High Density Exchange
IBM	International Business Machine
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
KVM	Keyboard Video Mouse
LCM2	Local Console Manager 2
MCS	Media Convergence Server
Mbps	Megabits per second
MFR1	Multi-Frequency Recommendation 1
MLPP	Multi-Level Precedence and Preemption
NFAS	Non-Facility Associated Signaling
PRI	Primary Rate Interface
Q.955.3	ISDN Signaling standard for E1 MLPP
RMK	Rack Monitor Keyboard
S/T	ISDN BRI 4-wire interface
SS7	Signaling System 7
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
U	ISDN BRI 2-wire interface
UCS	Universal Clock Synchronizer
USC	Universal Service Circuit
VoIP	Voice over Internet Protocol

Figure 2-3. SUT Test Configuration with ASLAN

9. SYSTEM CONFIGURATIONS. Table 2-2 provides the system configurations, hardware, and software components tested with the SUT. The SUT was tested in an operationally realistic environment to determine interoperability with a complement of DSN switches noted in Table 2-2. Table 2-2 lists the DSN switches which depict the tested configuration and is not intended to identify the only switches that are certified with the SUT. The SUT is certified with switching systems listed on the UC APL that offer the same certified interfaces.

Table 2-2. Tested System Configurations

DSN Switches			
System Name	**Software Release**		
Nortel CS2100	Succession Enterprise (SE)09.1		
Avaya S8710	Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)		
Siemens EWSD	19d with Patch Set 46		
Alcatel-Lucent 5ESS	5E16.2 Broadcast Warning Message (BWM) 08-0002		
REDCOM HDX V2.0a (R3P2) SUT Components			
MSU 0			
Part Number	**Part Description**		
MA0656-002	Controller		
MA0648-002	TSI		
MA0647-002	Announcer Board		
MA0530-322	BRI U-Interface 8 Circuit		
MA0060-005	Ring Generator		
MA0670-001	BRI S-Interface 8 circuit		
MA0706-001	Tone Service Board		
MA0728-163	Universal Clock Synchronizer Board		
MA0703-004	USC		
MA0473-163	UCS		
MA0609-310	DSP Service Board with Conf Module		
MA0724-301	Analog Line Board 16 circuit		
MA0317-904	Analog Line Circuit		
MA0708-115	Analog Line Circuit		
MA0702-302	Analog Line Circuit		
MA0602-201	Analog Line Circuit		
MA0683-144	E1/T1 MET 4 span		
MA0337-002	E1 Interface		
Cisco Unified CallManager Version 4.3(2) SR1b SUT Components			
Component (See note.)	**Release**	**Sub-Component (See note.)**	**Function**
CallManagers **MCS7825I3, MCS7835I2,** MCS7835H, MCS7835H2, MCS7825H3, MCS7835I1, MCS7825H2, MCS7835H1, MCS7825H, MCS7835I, MCS7825H1, MCS7825I1, MCS7845H, MCS7845H1, MCS7845H2, MCS7845I, MCS7845I1	4.3(2) SR1b	Not Applicable	Processing/Signaling
Cisco 3825 Integrated Services Router (Gateway)	IOS 12.4(9)T3	VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, Direct Inward Dial
		VWIC 2MFT T1	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1
		NM-HDV2-2T1/E1	2-port T1/E1 IP Communications HD Voice/fax NM, 2 T1/E1 Controllers
		NME-16ES-1G	Switch Module

Table 2-2. Tested System Configurations (continued)

Cisco Unified CallManager Version 4.3(2) SR1b SUT Components			
Component (See note.)	**Release**	**Sub-Component (See note.)**	**Function**
Cisco 3845 Integrated Services Router (Gateway)	IOS 12.4(9)T4	VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, Direct Inward Dial
		VWIC 2MFT T1	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1
		NM-HDV2-2T1/E1	2-port T1/E1 IP Communications HD Voice/fax NM, 2 T1/E1 Controllers
Cisco 3750-24TS, 3750-24PS-E, 3750-24FS-S, 3750-24PS-S, 3750-24TS-S, 3750-24TS-E, 3750-48PS-E, 3750-48PS-S, 3750-48TS-S, 3750-48TS-E, 3750G-48PS-E, 3750G-24T-S, 3750G-24T-E, 3750G-24TS-S, 3750G-24TS-E, 3750G-24TS-S1U, 3750G-24TS-E1U, 3750G-24PS-S, 3750G-24PS-E, 3750G-48TS-E, 3750G-48PS-S	12.2(40)SE	Not Applicable	System Connectivity
Cisco 3750 NM	12.2(35)SE5	NME-16ES-1G	Network Module

Local Management Terminals		
Hardware	**Firmware**	**Function**
BSI RMK-107KVM	V0.2	Management Workstation
IBM LCM2 KVM Switch	00.08.12.00	Provides connectivity to all the components connected to the KVM backplane

Telephone Instruments			
Interface Type	**Model**	**Release**	**Function**
2-Wire Analog	Panasonic KX-TS15-W	Not Applicable	Analog phone
ISDN BRI S/T and U	Tone Commander 6210T-B-02P and 6210-B-02H	01.07.22.01	ISDN BRI phone
VoIP	CP-7906G, CP-7911G, CP-7941G, CP-7942G, CP-7945G, CP-7961G, CP-7962G, CP-7965G, CP-7970G, CP-7971G-GE, CP-7975G	SCCP11.8-4-1SR1S	IP Phone (with push-to-talk handset or with standard handset)

NOTE: Components bolded and underlined were tested by JITC. The other components in the family series were not tested; however, they utilize the same IOS software and hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.

LEGEND:

5ESS	Class 5 Electronic Switching System	Mbps	Megabits per second
BR8	Basic Rate 8	MCS	Media Convergence Server
BRI	Basic Rate Interface	MFT	Multiflex Trunk
BSI	Broadax Systems Incorporated	MSU	Modular Switching Unit
CS	Communication Server	NM	Network Module
DID	Direct Inward Dial	RJ	Registered Jack
DSN	Defense Switched Network	RMK	Rack Monitor Keyboard
E1	European Basic Multiplex Rate (2.048 Mbps)	SUT	System Under Test
EWSD	Elektronisches Wählsystem Digital	S/T	ISDN BRI 4-wire interface
FXS	Foreign Exchange Station	TSI	Time Slot Interchange
HDV2	High Density Voice Fax	T1	Digital Transmission Link Level 1 (1.544 Mbps)
HDX	High Density Exchange	U	ISDN BRI 2-wire interface
IOS	Internetwork Operating System	USC	Universal Service Circuit
IP	Internet Protocol	VIC	Voice Interface Card
ISDN	Integrated Services Digital Network	VWIC	Voice WAN Interface Card
JITC	Joint Interoperability Test Command	VoIP	Voice over Internet Protocol
KVM	Keyboard Video Mouse	WAN	Wide Area Network

10. TESTING LIMITATIONS. NONE.

11. TEST RESULTS

a. Discussion

(1) DSN Trunk Interfaces. The SUT met all critical CRs and FRs for the following interfaces: T1 CAS with Dual Tone Multi-Frequency (DTMF), Dial Pulse (DP), and Multi-Frequency Recommendation 1 (MFR1) signaling; E1 CAS with DTMF, DP, and MFR1 signaling; T1 ISDN PRI National ISDN Standard 1 or 2 (NI 1/2) American National Standards Institute (ANSI) T1.619a; and E1 PRI ITU-T Q.955.3 with the minor exceptions noted in the subparagraph below:

(a) The SUT does not support Non-Facility Associated Signaling (NFAS) on the Cisco enclave ISDN PRI National ISDN Standard 2 (NI2) interface. NFAS is supported on the REDCOM HDX. Therefore, the REDCOM enclave must include at least one HDX shelf equipped with multiple ISDN PRI T1s. Both enclaves do support FAS.

(b) A discrepancy exists that is associated with the monitoring tool that Cisco enclave uses to check the status of the ISDN PRI trunks on the gateway. The monitoring tool occasionally provides an inaccurate representation of the status of the channels on the trunks when they are busied by the remote switching system. The SUT will occasionally provide an indication that the channel that was busied out by the far-end switch remains in an idle condition. This anomaly can be eliminated by insuring the trunks are busied at both the remote end and at the SUT. Furthermore, when this anomaly does occur, the correct busy state of the trunks is reflected in layer 3 protocol of the ISDN PRI interface, therefore, the operational impact is minor.

(2) DSN Line Interfaces. The SUT met all critical interoperability certification requirements for the following DSN line interfaces with a minor configuration change and the exceptions noted in the subparagraphs below: 2-wire analog, ISDN BRI NI1/2, and VoIP.

(a) A configuration change was required on the Cisco enclave analog gateways to meet the requirement for interoperability with secure devices, specifically the L3 Omni Secure Wireline Terminal. On the individual voice ports, the minimum and maximum settings for "timing hookflash in" had to be changed to a maximum value of 500 ms and a minimum value of 150 ms. Otherwise, a call that is placed between two Omni devices on the SUT will not disconnect when placed on hook.

(b) The conference disconnect tone that is provided by the REDCOM enclave does not meet the specifications designated in UCR, section 5.5.2. The SUT conference disconnect tone is distinguishable from other DSN tones and cadences; therefore, this anomaly has a minor operational impact.

(c) The precedence above ROUTINE ringing cadence that the REDCOM enclave applies to BRI phones does not meet the specifications as detailed in the UCR,

section 5.5.1. The precedence above ROUTINE cadence is distinct from the ROUTINE cadence when it is configured properly; therefore this anomaly has no operational impact.

(3) Features and Capabilities. The SUT met all critical interoperability certification requirements for Features and Capabilities.

(a) Common Features. The SUT met all critical CRs and FRs for common features with the minor exceptions noted in the subparagraph below:

1. The SUT does not support Call Pickup between the Cisco enclave and the REDCOM enclave. This solution is unique in that it offers SMEO functionality with two switches (Cisco and REDCOM) and each switch offers call pickup. This action is mitigated by not mixing call pickup groups between the two switch enclaves. DISA's adjudication of this discrepancy was ruled to have a minor operational impact.

2. When CFV is assigned to any station on the REDCOM enclave and CFV is invoked by the user, any station with CFV invoked does not receive a "ping" ring when calls are being forwarded. The operational impact is minor.

3. Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog stations of the Cisco enclave. Call Forward Variable, Three-way Calling, Call Hold, and Call Transfer are supported on VoIP stations only. These features are required for a SMEO for all instruments; however, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

4. Denied Originating Service is not supported by the SUT and is therefore not covered in this certification. This feature is not required for a SMEO.

5. The Cisco enclave does not support Call Waiting. However, there is no operational impact because the requirement is satisfied with multiple line appearances having a busy trigger. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

6. All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions.

7. Although the Cisco enclave does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP stations. This provides the ability for a user to receive additional calls while active with another call. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. There is no operational impact.

8. A short "ping" ring is not provided when calls are forwarded on the Cisco enclave; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact.

9. When a ROUTINE call is placed to a hunt group, and a ring-no-answer condition occurs, the calling party is diverted to the MLPP alternate directory number. This configuration must be done to allow correct treatment to be provided to precedence calls above ROUTINE that are placed to the hunt group. The UCR requires this only for precedence above ROUTINE calls. There is no operational impact.

(b) Attendant. The SUT does not support this feature. This is not a required feature for a SMEO. There is no risk associated with the SUT not supporting this feature.

(c) Public Safety. The SUT met all critical CRs and FRs.

(d) Conferencing. Met all critical CRs and FRs with the REDCOM enclave.

(e) Nailed-up Connections. This feature is not supported by the SUT. This is not a required feature for a SMEO. There is no risk associated with the SUT not supporting this feature.

(f) DSN Hotline Services. Met all critical CRs and FRs with the following minor exception: The SUT does not support the ISDN PRI code set 5 off hook indicator elements for hotline services as required by the UCR. The vendor began testing prior to 14 June 2008 and, therefore, was not required to provide this feature. This anomaly has minor operational impact.

(g) MLPP. The SUT met all critical CRs and FRs with the minor exceptions noted in the subparagraph below: The SUT met all critical CRs and FRs for the following preemption types: Preempt for reuse-answered, Preempt for reuse-unanswered, Preempt not for reuse-answered, and preempt not for reuse-unanswered.

1. The SUT does not support the Loss of C2 announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. This requirement is currently under review by DISA and the Joint Staff. In addition, the specific conditions that invoke this announcement have not yet been defined. As a result, the vendors are not required to be in compliance until 18 months from the date the requirement is fully defined.

2. The SUT does not support Method 1 preemption search algorithm if the trunks are a combination of the Cisco and REDCOM enclaves. In order to use the Method 1 search preemption search algorithm, all trunk groups must be member of the Cisco Gateway or the HDX switch. DISA's adjudication of this discrepancy was ruled to have a minor operational impact.

3. The SUT does not maintain the precedence level when transferring a call between the Cisco enclave and the REDCOM enclave. This discrepancy is due to the functionality between the Cisco and REDCOM enclaves. DISA's adjudication of this discrepancy was ruled to have a minor operational impact.

4. The SUT does not support Call Pickup between the Cisco enclave and the REDCOM enclave. This solution is unique in that it offers a SMEO functionality with two switches (Cisco and REDCOM) and each switch offers call pickup. This action is mitigated by not mixing call pickup groups between the two switch enclaves. DISA's adjudication of this discrepancy was ruled to have a minor operational impact.

5. When the initiator of a three-way call is preempted, the remaining parties do not receive a conference disconnect tone. However, the remaining members of the three-way call do stay connected. DISA's adjudication of this discrepancy was ruled to have a minor operational impact.

(h) Call Processing. The SUT met all critical CRs and FRs with the minor exceptions noted in the subparagraph below:

1. The SUT does not support the full complement of CoS tables as specified in the UCR. The SUT supports 255 CoS tables for analog lines and does not support CoS tables on access lines, number codes, trunks, or groups of trunks. This limitation has posed a minor operational impact within the DSN when assigning lines and trunks on the SUT.

2. The SUT does not support calling number delivery. This is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

(i) Network Management. Met all critical CRs and FRs with an Internet Protocol (IP) interface. To meet the GR-517-CORE call capacity requirements the Pacstar 6800 SMEO limitations for trunks and lines are as follows:

1. Trunks terminated on the Cisco enclave only

- Up to 30,000 VoIP Phones (Cisco enclave only)
- Up to 7,900 Analog phones (Cisco and REDCOM enclaves)
- 529 external T1s (Facility Associated Signaling only)
- Up to 71 T1 ISDN PRIs on the umbilical between the two enclaves

2. Trunks terminated on the REDCOM enclave only

- A combination of 11,000 VoIP and analog phones, or 16,000 phones terminated to the Cisco enclave only
- Up to 58 T1 ISDN PRIs on the umbilical between the two enclaves
- 71 T1s or 54 E1 interfaces (Maximum of 1614 time slots)

Note: One BRI phone can be substituted with two analog phones.

(j) ISDN Services. Met all critical CRs and FRs with the following minor exception: The SUT does not support Non-Facility Associated Signaling (NFAS) on the Cisco enclave ISDN PRI National ISDN Standard 2 (NI2) interface. NFAS is supported on the REDCOM HDX. Therefore, the REDCOM enclave must include at least one HDX shelf equipped with multiple ISDN PRI T1s. Both enclaves do support FAS.

(k) Synchronization. The SUT met all critical CRs and FRs.

(l) Reliability. The SUT met all critical CRs and FRs via the LoC. Backup power, power components, Uninterruptible Power Supply (UPS) requirements, UPS load capacity, and alarms are non-testable requirements. It is the responsibility of the respective base/post/camp/station communication agency to provide this with the SUT when installed.

(m) Security. Security is tested by DISA-led Information Assurance test teams and published in a separate report, reference (c).

(4) Network Gateways. The SUT met all critical interoperability certification requirements for the Public Switched Telephone Network (PSTN) Network Gateways. The SUT met all critical CRs and FRs for the following interfaces: T1 CAS with DTMF, DP, and MFR1 signaling; E1 CAS with DTMF, DP, and MFR1 signaling; T1 ISDN PRI NI 1/2 ANSI T1.607; and E1 PRI ITU-T Q.931 with the minor exceptions noted in the subparagraph below:

(a) The SUT does not support Non-Facility Associated Signaling (NFAS) on the Cisco enclave ISDN PRI National ISDN Standard 2 (NI2) interface. NFAS is supported on the REDCOM HDX. Therefore, the REDCOM enclave must include at least one HDX shelf equipped with multiple ISDN PRI T1s. Both enclaves do support FAS.

(b) A discrepancy exists that is associated with the monitoring tool that Cisco enclave uses to check the status of the ISDN PRI trunks on the gateway. The monitoring tool occasionally provides an inaccurate representation of the status of the channels on the trunks when they are busied by the remote switching system. The SUT will occasionally provide an indication that the channel that was busied out by the far-end switch remains in an idle condition. This anomaly can be eliminated by insuring the trunks are busied at both the remote end and at the SUT. Furthermore, when this anomaly does occur, the correct busy state of the trunks is reflected in layer 3 protocol of the ISDN PRI interface, therefore, the operational impact is minor.

(5) VoIP. The SUT is certified with any ASLAN on the UC APL.

(a) VoIP System. The UCR, appendix 3, section A3.2, outlines the requirements for the VoIP system. The VoIP system requirements encompass end-to-end VoIP requirements. The following paragraphs detail the results of the SUT VoIP solution.

1. Voice Quality. In accordance with the UCR, appendix 3, section A3.2.1, VoIP calls shall have an average Mean Opinion Score (MOS) of at least 4.0 as measured in accordance with ITU-T P.800 voice quality standards. This applies from handset to handset and from handset to gateway trunk in the DSN. For intra-switch calls, the SUT VoIP solution had an average MOS of 4.34 with a minimum measured MOS value of 4.09. The average inter-switch MOS was 4.36 with a minimum measured MOS value of 4.12. This average was based on a total of 910 calls. Additionally, VoIP systems shall not lose more than 150 ms of voice media in any five-minute period. This applies from handset to handset and from handset to gateway trunk to the DSN. The SUT met this requirement with a loss of no more than 0.0 ms of voice media packets in any five-minute period.

2. Codec. In accordance with the UCR, appendix 3, section A3.2.2, the ITU-T G.711 Pulse Code Modulation (PCM) CODEC with a 20 ms packet fill was required and was met by the SUT VoIP solution.

3. Multi-Level Precedence and Preemption (MLPP). In accordance with the UCR, appendix 3, section A3.2.3, the VoIP system shall meet all MLPP requirements identified in UCR, section 3. All critical MLPP features and functions were met by the SUT.

4. Security. Security requirements in accordance with the UCR, appendix 3, section A3.2.4, are verified using the Information Assurance Test Plan. Results of the security testing are reported in a separate test report generated by the DISA Information Assurance test personnel, reference (c).

5. Network Management (NM). In accordance with the UCR, appendix 3, section A3.2.5, the vendor is required to provide a management system to monitor the performance of the ASLAN portion of the VoIP system. This requirement was verified via a LoC because of the numerous third party systems and applications capable of performing this function. The switching system NM requirements in accordance with the UCR, section 9, are not required for a PBX 1 and were not tested.

6. Synchronization. In accordance with the UCR, appendix 3, section A3.2.6, the VoIP system shall meet all synchronization requirements identified in UCR, section 11. The SUT derived synchronization with line timing mode via traditional T1 TDM-based interfaces.

7. Latency. The UCR, appendix 3, section A3.2.7, states that one-way system latency for the VoIP system must be 60 ms or less as averaged over any five-minute period. The latency requirement is measured from IP handset to the egress

trunk. The SUT average latency over 910 inter-switch calls, with a minimum duration of 5 minutes for each call, was measured to be 54.69 ms.

8. Internet Protocol version 6 (IPv6). An IPv6 capable system or product, as defined in the UCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of Internet Protocol version 4 (IPv4). IPv6 capability is currently satisfied by a vendor LoC signed by the Vice President of their respective company. The vendor stated, in writing, compliance to the following criteria:

a. Conformant with IPv6 standards profile contained in the Department of Defense Information Technology Standards Registry (DISR). These standards are delineated in the UCR, appendix 11.

b. Maintaining interoperability in heterogeneous environments and with IPv4.

c. Commitment to upgrade as the IPv6 standard evolves.

d. Availability of contractor/vendor IPv6 technical support.

The SUT was tested with IPv4 only. In accordance with the Office of Secretary IPv6 Rules of engagement a solution can be tested and certified for IPv4 only, however the vendor is required to stipulate in an IPv6 LoC their way ahead to be IPv6 capable by end of CY 2008. In addition the vendor is required to return for retest with this IPv6 solution prior to the end of CY 2008. The vendor stated in their IPv6 LoC submission that they will not be able to deliver an IPv6 capable solution until 31 May of 2010. The vendor received a waiver for this requirement from OSD on 9 March 2009.

9. In accordance with the UCR, appendix 3, section A3.2.9.1, the VoIP system components shall meet the following requirements:

a. All components shall be capable of implementing Service Class tagging using the 6-bit Differentiated Services Code Points (DSCPs) field in the IP header. The SUT end instruments used 6-bit service class tagging in the IP header, which meets the requirement.

b. All components shall be capable of assigning DSCP (0-63) to any distinct service class for traffic that traverses the device in accordance with UCR, Tables A3-1 and A3-2. The VoIP SUT solution assigned DSCP values of 48 for signaling and 46 for voice media, which meets the requirement.

c. Any component that supports Real Time traffic and data shall be capable of tagging all Real Time traffic with an Institute of Electrical and Electronics Engineers (IEEE) 802.1Q 2-byte Tag Control Information (TGI) field 12-bit virtual LAN

(VLAN) Identification (VID). The VoIP SUT solution supports Real Time traffic. Data was not mixed with Real Time traffic, so tagging was conditional.

10. In accordance with the UCR, appendix 3, section A3.2.9.2, the VoIP system end user devices shall meet the following requirements:

a. All end instrument components shall be capable of implementing Service Class tagging using the 6-bit DSCPs field in the IP header. The SUT end instruments used 6-bit service class tagging in the IP header, which meets the requirement.

b. The DSCPs shall be assigned to any distinct service class that originates or traverses the end instrument. The DSCPs may be assigned by either having the end instrument itself assign the DSCP to the distinct service class or having the call control portion of the VoIP system tell the end instrument what DSCP to insert to the distinct service class. The SUT end instrument assigned a DSCP value of 48 for voice signaling and 46 for voice media, which meets the requirement.

c. Any end instrument that supports Real Time traffic shall be capable of tagging all Real Time traffic with an IEEE 802.1Q 2-byte TCI field 12-bit VID. The SUT tagged the voice VID with 114 and the data VID with 11, which meets the requirement. The Cisco VoIP phones that met the critical interoperability requirements for certification were the CP7906G, CP7911G, CP7941G, CP7942G, CP7945G, CP7961G, CP7962G, CP7965G, CP7970G, CP7971G-GE, and CP7975G. The above phones have been tested and are certified for 100 Mbps shared access (i.e., same switch port is shared by PC and IP phone) with the exception of the CP7906G. The CP7906G phone does not support shared access. The following phones were tested and are certified for 1 gig shared access: CP7975G, CP7965G, and CP7945G. The CP7971G-GE phone is capable of web browsing; however, this feature was not tested, is not covered by this certification, and is not authorized for use within the DSN. All VoIP phones were tested using Secure Real Time Protocol (SRTP) which encrypts the media stream. The SRTP is able to encrypt only IP phone to IP phone intra-switch traffic and IP phone to gateway intra-switch traffic. All other calls (i.e. analog to analog, or analog to gateway traffic) are not encrypted.

11. In accordance with the UCR, appendix 3, section A3.2.10, the VoIP system shall meet the maximum downtime of 80 minutes per year for the system and 120 minutes per year for the subscriber. This requirement was met via a LoC.

(b) Scalability. The MCS7835s can support a maximum of 2,500 IP subscribers, the MCS7825 can support 1,000 IP subscribers, and the MCS7845 can support 7,500 IP subscribers. However; the configurations range from 1,000 IP subscribers with two MCS7825s to 30,000 subscribers with eight MCS7845s. The recommendation is to consult an engineer to determine the appropriate configurations. The SUT is certified with any certified ASLAN on the UC APL. The ASLAN can be scaled to meet the maximum subscribers as long as it is comprised of the equipment

and software listed in this certification, and meets the traffic engineering constraints contained in the UCR, appendix 3.

b. System Interoperability Results. The SUT is certified for joint use in the DSN as a SMEO in accordance with the requirements set forth in the UCR. The identified test discrepancies that remained open after software patches were applied and regression testing was completed have an overall minor operational impact. The following CallManagers were not tested; however, they utilize the same IOS software and hardware: MCS7835H, MCS7835H2, MCS7825H3, MCS7835I1, MCS7825H2, MCS7835H1, MCS7825H, MCS7835I, MCS7825H1, MCS7825I1, MCS7845H, MCS7845H1, MCS7845H2, MCS7845I, and MCS7845I1. The following Catalyst 3750 series switches were not tested; however they utilize the same IOS and software and hardware: 3750-24PS-E, 3750-24FS-S, 3750-24PS-S, 3750-24TS-S, 3750-24TS-E, 3750-48PS-E, 3750-48PS-S, 3750-48TS-S, 3750-48TS-E, 3750G-48PS-E, 3750G-24T-S, 3750G-24T-E, 3750G-24TS-S, 3750G-24TS-E, 3750G-24TS-S1U, 3750G-24TS-E1U, 3750G-24PS-S, 3750G-24PS-E, 3750G-48TS-E, and 3750G-48PS-S. JITC analysis determined them to be functionally identical for interoperability certification purposes. The SUT interoperability test summary is shown in Table 2-3. The SUT Interoperability Requirements/Status is shown in Table 2-4.

Table 2-3. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, DP)	Yes	Certified	Met all critical CRs and FRs with the REDCOM enclave. Although Cisco offers this interface, it is not certified.
T1 CAS (MFR1)	No	Certified	Met all critical CRs and FRs with the REDCOM enclave. Although Cisco offers this interface, it is not certified.
E1 CAS (DTMF, DP)	Yes (Europe only)	Certified	Met all critical CRs and FRs with the REDCOM enclave. Although Cisco offers this interface, it is not certified.
E1 CAS (MFR1)	No (Europe only)	Certified	Met all critical CRs and FRs with the REDCOM enclave. Although Cisco offers this interface, it is not certified.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the REDCOM enclave. The Cisco enclave does not support NFAS.[1] The Cisco enclave monitoring tool occasionally provides inaccurate reports when a remote trunk is busy.[2]
E1 PRI (ITU-T Q.955.3)	No (Europe only)	Certified	Met all critical CRs and FRs with the REDCOM enclave. The E1 ISDN PRI interface is supported by the Cisco enclave; however, it does not support ITU-T Q.955.3 MLPP.
T1 SS7 (ANSI T1.619a)	No	Not Tested	T1 SS7 is supported by the SUT; however it was not tested. The SUT T1 SS7 interface is therefore not certified by JITC. This is not a required interface for a SMEO.
E1 SS7 (ANSI T1.619a)	No	Not Tested	E1 SS7 is not supported by the SUT. This is not a required interface for a SMEO. There is no risk associated with the SUT not supporting this interface.
2-wire/4-wire E&M	No	Not Tested	2-wire/4-wire E&M is supported by the SUT; however it was not tested. The SUT 2-wire/4-wire E&M interface is therefore not certified by JITC. This is not a required interface for a SMEO.

Table 2-3. SUT Interoperability Test Summary (continued)

DSN Line Interfaces				
Interface & Signaling		**Critical**	**Status**	**Remarks**
2-Wire Analog (GR-506-CORE)		Yes	Certified	Met all critical CRs and FRs with a minor configuration change[3] and the following minor exceptions: The REDCOM enclave conference disconnect tone on phones connected to the HDX switch do not meet the specifications.[4]
ISDN BRI NI 1/2		Yes	Certified	Met all critical CRs and FRs with the REDCOM enclave with the following minor exceptions: The conference disconnect tone does not meet the specifications.[4] The precedence above ROUTINE ringing cadence that the SUT applies to BRI phones does not meet the specifications.[5] This interface is not supported on the Cisco enclave.
2-Wire Proprietary Digital		No	Not Tested	2-Wire Proprietary Digital is not supported by the SUT. This is not a required interface for a SMEO. There is no risk associated with the SUT not supporting this interface.
VoIP (Session Initiation Protocol)		No	Certified	Met all critical CRs and FRs with the Cisco enclave.
Common Features		Yes	Certified	Met all critical CRs and FRs for the with the following minor exception: The SUT does not support Call Pickup between the two enclaves.[6] The REDCOM enclave does not correctly support the call forwarding variable "ping" ring feature.[7] Met all critical CRs and FRs for the Cisco enclave with the following minor exceptions: Full compliance of DSN Common Call Features was not met.[8, 9, 10, 11, 12, 13]
Attendant		No	Not Tested	The SUT does not support this feature. This is not a required feature for a SMEO. There is no risk associated with the SUT not supporting this feature.
Public Safety		Yes	Certified	Met all critical CRs and FRs.
Conferencing	Preset	No	Certified	Met all critical CRs and FRs with the REDCOM enclave.
	Meet-me	Yes	Certified	Met all critical CRs and FRs with the REDCOM enclave.
	Progressive	No	Certified	Met all critical CRs and FRs with the REDCOM enclave.
Nailed-up Connections		No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a SMEO. There is no risk associated with the SUT not supporting this feature.
DSN Hotline Services		Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT does not support the ISDN PRI code set 5 off hook indicator elements for hotline services.[14]
MLPP		Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support the loss of Command and Control announcement.[15] The SUT does not support Method 1 preemption search algorithim if the trunks are a combination of Cisco and REDCOM enclaves.[16] The SUT does not maintain the precedence level when transferring a call from the Cisco enclave to the REDCOM enclave.[17] The SUT does not support Call Pickup between the Cisco enclave and the REDCOM enclave.[6] When the initiator of a three-way call is preempted, the remaining parties do not receive a conference disconnect tone.[18]
Call Processing		Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The REDCOM enclave does not support the full complement of CoS tables.[19] The SUT does not support calling number delivery.[20]
Network Management		Yes	Certified	Met all critical CRs and FRs with Internet Protocol (IP) interfaces.
ISDN Services		Yes	Certified	Met all critical CRs and FRs. The Cisco enclave does not support NFAS. NFAS is supported on the REDCOM enclave.[1] The operational impact is minor.
Synchronization		Yes	Certified	Met all critical CRs and FRs.
Reliability		Yes	Certified	Met all critical CRs and FRs.[21]
Security		Yes	Certified	See note 22.
VoIP System		No	Certified	The SUT is certified for VoIP with any certified ASLAN posted on the UC APL. See notes 23 and 24.

Table 2-3. SUT Interoperability Test Summary (continued)

Network Gateways				
Gateway	**Interface & Signaling**	**Critical**	**Status**	**Remarks**
PSTN	T1 CAS (DTMF, DP, MFR1)	Yes	Certified	Met all critical CRs and FRs with the REDCOM enclave. Although Cisco offers this interface, it is not certified.
	E1 CAS (DTMF, DP, MFR1)	Yes (Europe only)	Certified	Met all critical CRs and FRs with the REDCOM enclave. Although Cisco offers this interface, it is not certified.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	Yes	Certified	Met all critical CRs and FRs with the REDCOM enclave. The Cisco enclave does not support NFAS.[1] The operational impact is minor.
	E1 PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.
	Ground Start Line	Yes	Certified	Met all critical CRs and FRs.

NOTES:

1 The SUT does not support NFAS on the Cisco enclave ISDN PRI National ISDN Standard 2 (NI2) interface. NFAS is supported on the REDCOM HDX. Therefore, the REDCOM enclave must include at least one HDX shelf equipped with multiple ISDN PRI T1s. Both enclaves do support FAS.

2 A discrepancy exists that is associated with the monitoring tool that Cisco enclave uses to check the status of the ISDN PRI trunks on the gateway. The monitoring tool occasionally provides an inaccurate representation of the status of the channels on the trunks when they are busied by the remote switching system. The SUT will occasionally provide an indication that the channel that was busied out by the far-end switch remains in an idle condition. This anomaly can be eliminated by insuring the trunks are busied at both the remote end and at the SUT. Furthermore, when this anomaly does occur, the correct busy state of the trunks is reflected in layer 3 protocol of the ISDN PRI interface, therefore, the operational impact is minor.

3 A configuration change was required on the Cisco enclave analog gateways to meet the requirement for interoperability with secure devices, specifically the L3 Omni Secure Wireline Terminal. On the individual voice ports, the minimum and maximum settings for "timing hookflash in" had to be changed to a maximum value of 500 ms and a minimum value of 150 ms. Otherwise, a call that is placed between two Omni devices on the SUT will not disconnect when placed on hook.

4 The conference disconnect tone that is provided by the REDCOM enclave does not meet the specifications designated in UCR, section 5.5.2. The SUT conference disconnect tone is distinguishable from other DSN tones and cadences; therefore, this anomaly has a minor operational impact.

5 The precedence above ROUTINE ringing cadence that the REDCOM enclave applies to BRI phones does not meet the specifications as detailed in the UCR, section 5.5.1. The precedence above ROUTINE cadence is distinct from the ROUTINE cadence when it is configured properly; therefore this anomaly has no operational impact.

6 The SUT does not support Call Pickup between the Cisco enclave and the REDCOM enclave. This solution is unique in that it offers SMEO functionality with two switches (Cisco and REDCOM) and each switch offers call pickup. This action is mitigated by not mixing call pickup groups between the two switch enclaves. DISA's adjudication of this discrepancy was ruled to have a minor operational impact.

7 When CFV is assigned to any station on the REDCOM enclave and CFV is invoked by the user, any station with CFV invoked does not receive a "ping" ring when calls are being forwarded. The operational impact is minor.

8 Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog stations of the Cisco enclave. Call Forward Variable, Three-way Calling, Call Hold, and Call Transfer are supported on VoIP stations only. These features are required for a SMEO for all instruments; however, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. Denied Originating Service is not supported by the SUT and is therefore not covered in this certification. This feature is not required for a SMEO.

9 The Cisco enclave does not support Call Waiting. However, there is no operational impact because the requirement is satisfied with multiple line appearances having a busy trigger. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

10 All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions.

11 Although the Cisco enclave does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP stations. This provides the ability for a user to receive additional calls while active with another call. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. There is no operational impact.

12 A short "ping" ring is not provided when calls are forwarded on the Cisco enclave; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact.

13 When a ROUTINE call is placed to a hunt group, and a ring-no-answer condition occurs, the calling party is diverted to the MLPP alternate directory number. This configuration must be done to allow correct treatment to be provided to precedence calls above ROUTINE that are placed to the hunt group. The UCR requires this only for precedence above ROUTINE calls. There is no operational impact.

14 The SUT does not support the ISDN PRI code set 5 off hook indicator elements for hotline services as required by the UCR. The vendor began testing prior to 14 June 2008 and, therefore, was not required to provide this feature. This anomaly has minor operational impact.

15 The SUT does not support the Loss of C2 announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. This requirement is currently under review by DISA and the Joint Staff. In addition, the specific conditions that invoke this announcement have not yet been defined. As a result, the vendors are not required to be in compliance until 18 months from the date the requirement is fully defined.

16 The SUT does not support Method 1 preemption search algorithm if the trunks are a combination of the Cisco and REDCOM enclaves. In order to use the Method 1 search preemption search algorithm, all trunk groups must be member of the Cisco Gateway or the HDX switch. DISA's adjudication of this discrepancy was ruled to have a minor operational impact.

17 The SUT does not maintain the precedence level when transferring a call between the Cisco enclave and the REDCOM enclave. This discrepancy is due to the functionality between the Cisco and REDCOM enclaves. DISA's adjudication of this discrepancy was ruled to have a minor operational impact.

18 When the initiator of a three-way call is preempted, the remaining parties do not receive a conference disconnect tone. However, the remaining members of the three-way call do stay connected. DISA's adjudication of this discrepancy was ruled to have a minor operational impact.

19 The SUT does not support the full complement of CoS tables as specified in the UCR. The SUT supports 255 CoS tables for analog lines and does not support CoS tables on access lines, number codes, trunks, or groups of trunks. This limitation has posed a minor operational impact within the DSN when assigning lines and trunks on the SUT.

Table 2-3. SUT Interoperability Test Summary (continued)

NOTES (continued):

20 This is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

21 Backup power, power components, UPS requirements, UPS load capacity and alarms are non-testable requirements. It is the responsibility of the respective base/post/camp/station communication agency to provide this with the SUT when installed.

22 Security is tested by DISA-led Information Assurance test teams and published in a separate report, reference (c).

23 An IPv6 capable system or product, as defined in the UCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of their respective company. The vendor stated in writing, their intent to return to JITC for testing of their solution with IPv6 enabled earliest date available. In addition they stated in writing, compliance to the following criteria:

a. Conformant with IPv6 standards profile contained in the Department of Defense Information Technology Standards Registry (DISR). These standards are delineated in the UCR, appendix 11.

b. Maintaining interoperability in heterogeneous environments and with IPv4.

c. Commitment to upgrade as the IPv6 standard evolves.

d. Availability of contractor/vendor IPv6 technical support.

24 The SUT was tested with IPv4 only. In accordance with the Office of Secretary IPv6 Rules of engagement a solution can be tested and certified for IPv4 only, however the vendor is required to stipulate in an IPv6 LoC their way ahead to be IPv6 capable by end of CY 2008. In addition the vendor is required to return for retest with this IPv6 solution prior to the end of CY 2008. The vendor stated in their IPv6 LoC submission that they will not be able to deliver an IPv6 capable solution until 31 May of 2010. The vendor received a waiver for this requirement from OSD on 9 March 2009.

LEGEND:

ANSI	American National Standards Institute	GR-506-CORE	LSSGR: Signaling for Analog Interfaces	OSD	Office of the Secretary of Defense
APL	Approved Products List	HDX	High Density Exchange	PRI	Primary Rate Interface
ASLAN	Assured Services Local Area Network	IPv4	Internet Protocol version 4	PSTN	Public Switched Telephone Network
BRI	Basic Rate Interface	IPv6	Internet Protocol version 6	Q.931	Signaling Standard for ISDN
C2	Command and Control	ISDN	Integrated Services Digital Network	Q.955.3	ISDN signaling standard for E1 MLPP
CAS	Channel Associated Signaling	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	SMEO	Small End Office
CFV	Call Forwarding Variable			SS7	Signaling System 7
CoS	Class of Service	JITC	Joint Interoperability Test Command	SUT	System Under Test
CRs	Capability Requirements			T1	Digital Transmission Link Level 1 (1.544 Mbps)
CY	Calendar Year	LoC	Letters of Compliance	T1.607	ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
DISA	Defense Information Systems Agency	LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements		
DP	Dial Pulse				
DSN	Defense Switched Network	Mbps	Megabits per second		
DSS1	Digital Subscriber Signaling 1	MFR1	Multi-Frequency Recommendation 1	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DTMF	Dual Tone Multi-Frequency				
E&M	Ear and Mouth	MLPP	Multi-Level Precedence and Preemption	UC	Unified Capabilities
E1	European Basic Multiplex Rate (2.048 Mbps)	ms	milliseconds	UCR	Unified Capabilities Requirements
FAS	Facility Associated Signaling	NFAS	Non-Facility Associated Signaling	UPS	Uninterruptible Power Supply
FRs	Feature Requirements				
GR	Generic Requirement	NI 1/2	National ISDN Standard 1 or 2	VoIP	Voice over Internet Protocol

12. TEST AND ANALYSIS REPORT. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet), or http://199.208.204.125 (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi.

Table 2-4. SUT Interoperability Requirements/Status

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
T1 CAS (MFR1, DTMF, DP)	No	Certified (See note 1.)	Trunking	Direct Inward Dialing (C)	UCR Section 2.3.2	Met	
				Normal Wink Start Operations (R)	UCR Section 5.3.3.1.1	Met	
				Glare Operation (R)	UCR Section 5.3.3.1.2	Met	
				Abnormal Wink Start (R)	UCR Section 5.3.3.2.1	Met	
				Glare Resolution (R)	UCR Section 5.3.3.2.2	Met	
				Call for Service Timing (R)	UCR Section 5.3.5	Met	
				Guard Timing (R)	UCR Section 5.3.6	Met	
				Satellite Timing (R)	UCR Section 5.3.7	Met	
				Disconnect Control (R)	UCR Section 5.3.8	Met	
				Reselect and Retrial (R)	UCR Section 5.3.9	Met	
				Off-Hook Supervision Transition (R)	UCR Section 5.3.10	Met	
				Dial-Pulse Signals (R)	UCR Section 5.4.1	Met	
				DTMF Signaling (R)	UCR Section 5.4.2	Met	
				Standard Digit Format for Precedence (C)	UCR Section 5.4.2.1	Met	
				MFR1 2/6 Signaling (C)	UCR Section 5.4.3	Met	
				Alerting Signals and Tones (R)	UCR Section 5.5	Met	
				PCM-24 Digital Trunk Interface (R)	UCR Section 7.1	Met	
				Interface Characteristics (R)	UCR Section 7.1.1	Met	
				Supervisory Channel Associated Signaling (R)	UCR Section 7.1.2	Met	
				Clear Channel Capability (R)	UCR Section 7.1.3	Met	
				Alarm and Restoral Requirements (R)	UCR Section 7.1.4	Met	
				Interoperation of PCM-24 and PCM-30 (R)	UCR Section 7.3	Met	
				Integrated Digital Loop Carrier (C)	UCR Section 7.5	Not Tested	See note 2.
				Local Office Test Line (C)	UCR Section 2.5.1	Not Tested	See note 2.
				Outside Plant Test Lines (C)	UCR Section 2.5.2	Not Tested	See note 2.
				Test Incoming Trunks Tandem or Local State (C)	UCR Section 2.5.3	Not Tested	See note 2.
				Manual Test of Trunks (R)	UCR Section 2.5.4.2	Met	
				Trunk Group-Remove from Service (R)	UCR Section 2.5.5	Met	
				Trunk Group-Restore to Service (C)	UCR Section 2.5.6	Met	
				Carrier Group Alarm (R)	UCR Section 2.5.7	Met	
				Software Carrier Group Alarm (C)	UCR Section 2.5.7.1	Not Tested	See note 2.

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
T1 CAS (MFR1, DTMF, DP) (continued)	No	Certified (See note 1.)	Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				56 kbps switched data (R: PRI only)	UCR Section 3.10	Met	
				NX56 synchronous BER (R: PRI only)	UCR Section 3.10	Met	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
E1 CAS (MFR1, DTMF, DP)	Yes (Europe only)	Certified (See note 1.)	Trunking	Direct Inward Dialing (C)	UCR Section 2.3.2	Met	
				Normal Wink Start Operations (R)	UCR Section 5.3.3.1.1	Met	
				Glare Operation (R)	UCR Section 5.3.3.1.2	Met	
				Abnormal Wink Start (R)	UCR Section 5.3.3.2.1	Met	
				Glare Resolution (R)	UCR Section 5.3.3.2.2	Met	
				Call for Service Timing (R)	UCR Section 5.3.5	Met	
				Guard Timing (R)	UCR Section 5.3.6	Met	
				Satellite Timing (R)	UCR Section 5.3.7	Met	
				Disconnect Control (R)	UCR Section 5.3.8	Met	
				Reselect and Retrial (R)	UCR Section 5.3.9	Met	
				Off-Hook Supervision Transition (R)	UCR Section 5.3.10	Met	
				Dial-Pulse Signals (R)	UCR Section 5.4.1	Met	
				DTMF Signaling (R)	UCR Section 5.4.2	Met	
				Standard Digit Format for Precedence (C)	UCR Section 5.4.2.1	Met	
				MFR1 2/6 Signaling (C)	UCR Section 5.4.3	Met	
				Alerting Signals and Tones (R)	UCR Section 5.5	Met	
				PCM-30 Digital Trunk Interface (R)	UCR Section 7.2	Met	
				Interoperation of PCM-24 and PCM-30 (R)	UCR Section 7.3	Met	
				Integrated Digital Loop Carrier (C)	UCR Section 7.5	Not Tested	See note 2.
				Local Office Test Line (C)	UCR Section 2.5.1	Not Tested	See note 2.
				Outside Plant Test Lines (C)	UCR Section 2.5.2	Not Tested	See note 2.
				Test Incoming Trunks Tandem or Local State (C)	UCR Section 2.5.3	Not Tested	See note 2.
				Manual Test of Trunks (R)	UCR Section 2.5.4.2	Met	
				Trunk Group-Remove from Service (R)	UCR Section 2.5.5	Met	
				Trunk Group-Restore to Service (R)	UCR Section 2.5.6	Met	
				Carrier Group Alarm (R)	UCR Section 2.5.7	Met	
				Software Carrier Group Alarm (C)	UCR Section 2.5.7.1	Not Tested	See note 2.
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
E1 CAS (MFR1, DTMF, DP) (continued)	Yes (Europe only)	Certified (See note 1.)	Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				56 kbps switched data (R: PRI only)	UCR Section 3.10	Met	
				64 kbps switched data (R: PRI only)	UCR Section 3.10	Met	
				NX56 synchronous BER (R: PRI only)	UCR Section 3.10	Met	
				NX64 synchronous BER (R: PRI only)	UCR Section 3.10	Met	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	No	Certified	Trunking	Direct Inward Dialing (C)	UCR Section 2.3.2	Met	
				National ISDN 1/2 Primary Access (R)	UCR Section 2.3.4.1	Met	
				ISDN ANSI MLPP Service Capability (R)	UCR Section 2.3.4.1.1	Met	
				Alerting Signals and Tones (R)	UCR Section 5.5	Met	
				DSN ISDN User-to-Network Signaling (R)	UCR Section 5.7.1	Met	
				Application (R)	UCR Section 5.7.1.1	Met	See note 3.
				Physical Layer (R)	UCR Section 5.7.1.2	Met	
				Data Link Layer (R)	UCR Section 5.7.1.3	Met	
				Data Link Connection (R)	UCR Section 5.7.1.3.1	Met	
				Peer-to-Peer Procedures of Data-Link Layer (R)	UCR Section 5.7.1.3.2	Met	
				Layer 3 DSN User-to-Network Signaling (R)	UCR Section 5.7.1.4	Met	See note 4.
				DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)	UCR Section 5.7.1.4.2	Met	
				Sequence of Messages for DSN Circuit-Switched Calls (R)	UCR Section 5.7.1.4.3	Met	
				Message Functional Definition and Content (R)	UCR Section 5.7.1.4.4	Met	
				General Message Format and Information Elements Coding (R)	UCR Section 5.7.1.4.5	Met	
				Supplementary Services (C)	UCR Section 5.7.1.4.6	Not Tested	See note 2.
				PCM-24 Digital Trunk Interface (R)	UCR Section 7.1	Met	
				Interface Characteristics (R)	UCR Section 7.1.1	Met	
				Clear Channel Capability (R)	UCR Section 7.1.3	Met	
				Alarm and Restoral Requirements (R)	UCR Section 7.1.4	Met	
				Interoperation of PCM-24 and PCM-30 (R)	UCR Section 7.3	Met	
				Integrated Digital Loop Carrier (C)	UCR Section 7.5	Not Tested	See note 2.
				Local Office Test Line (C)	UCR Section 2.5.1	Not Tested	See note 2.
				Outside Plant Test Lines (C)	UCR Section 2.5.2	Not Tested	See note 2.
				Test Incoming Trunks Tandem or Local State (C)	UCR Section 2.5.3	Not Tested	See note 2.
				Manual Test of Trunks (R)	UCR Section 2.5.4.2	Met	
				Trunk Group-Remove from Service (R)	UCR Section 2.5.5	Met	
				Trunk Group-Restore to Service (R)	UCR Section 2.5.6	Met	
				Carrier Group Alarm (R)	UCR Section 2.5.7	Met	
				Software Carrier Group Alarm (C)	UCR Section 2.5.7.1	Not Tested	See note 2.

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
T1 ISDN PRI NI 1/2 (ANSI T1.619a) (continued)	No	Certified	Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				56 kbps switched data (R: PRI only)	UCR Section 3.10	Met	
				64 kbps switched data (R: PRI only)	UCR Section 3.10	Met	
				NX56 synchronous BER (R: PRI only)	UCR Section 3.10	Met	
				NX64 synchronous BER (R: PRI only)	UCR Section 3.10	Met	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
			VTC	ITU-T H.320 (R: PRI only)	FTR 1080B-2002	Met	

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
E1 ISDN PRI (ITU-T Q.955.3)	No	Certified (See note 5.)	Trunking	Direct Inward Dialing (C)	UCR Section 2.3.2	Met	
				ITU-T ISDN Primary Access (C)	UCR Section 2.3.4.2	Met	
				ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (C)	UCR Section 2.3.4.2.1	Met	
				DSN ISDN User-to-Network Signaling (R)	UCR Section 5.7.1	Met	
				Application (R)	UCR Section 5.7.1.1	Met	
				Physical Layer (R)	UCR Section 5.7.1.2	Met	
				Data Link Layer (R)	UCR Section 5.7.1.3	Met	
				Data Link Connection (R)	UCR Section 5.7.1.3.1	Met	
				Peer-to-Peer Procedures of Data-Link Layer (R)	UCR Section 5.7.1.3.2	Met	
				Layer 3 DSN User-to-Network Signaling (R)	UCR Section 5.7.1.4	Met	
				DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)	UCR Section 5.7.1.4.2	Met	
				Sequence of Messages for DSN Circuit-Switched Calls (R)	UCR Section 5.7.1.4.3	Met	
				Message Functional Definition and Content (R)	UCR Section 5.7.1.4.4	Met	
				General Message Format and Information Elements Coding (R)	UCR Section 5.7.1.4.5	Met	
				Supplementary Services (C)	UCR Section 5.7.1.4.6	Not Tested	See note 2.
				PCM-30 Digital Trunk Interface (R)	UCR Section 7.2	Met	
				Interoperation of PCM-24 and PCM-30 (R)	UCR Section 7.3	Met	
				Integrated Digital Loop Carrier (C)	UCR Section 7.5	Not Tested	See note 2.
				Local Office Test Line (C)	UCR Section 2.5.1	Not Tested	See note 2.
				Outside Plant Test Lines (C)	UCR Section 2.5.2	Not Tested	See note 2.
				Test Incoming Trunks Tandem or Local State (C)	UCR Section 2.5.3	Not Tested	See note 2.
				Manual Test of Trunks (R)	UCR Section 2.5.4.2	Met	
				Trunk Group-Remove from Service (R)	UCR Section 2.5.5	Met	
				Trunk Group-Restore to Service (R)	UCR Section 2.5.6	Met	
				Carrier Group Alarm (R)	UCR Section 2.5.7	Met	
				Software Carrier Group Alarm (C)	UCR Section 2.5.7.1	Not Tested	See note 2.

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
E1 ISDN PRI (ITU-T Q.955.3) (continued)	No	Certified (See note 5.)	Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				56 kbps switched data (R: PRI only)	UCR Section 3.10	Met	
				64 kbps switched data (R: PRI only)	UCR Section 3.10	Met	
				NX56 synchronous BER (R: PRI only)	UCR Section 3.10	Met	
				NX64 synchronous BER (R: PRI only)	UCR Section 3.10	Met	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
			VTC	ITU-T H.320 (R: PRI only)	FTR 1080B-2002	Met	

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Line Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
2-Wire Analog	Yes	Certified	Access	Directory Number Identification (R)	UCR Section 2.1.1	Met	
				PBX Line (C)	UCR Section 2.3.1	Met	
				Analog Line (R)	UCR Section 2.3.5	Met	
				Basic Line Test Capabilities (R)	UCR Section 2.5.4.1.1	Met	
				Advanced Line Test Capabilities (C)	UCR Section 2.5.4.1.2	Not Tested	See note 2.
				Network Power Systems for External Interfaces (R)	UCR Section 5.1	Met	
				Loop Start Line (R: 2-Wire Analog only)	UCR Section 5.2.1	Met	
				Reverse Battery (R)	UCR Section 5.3.1	Met	
				Alerting Signals and Tones (R)	UCR Section 5.5	Met	See note 6.
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	See note 7.
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Certified	Access	Directory Number Identification (R)	UCR Section 2.1.1	Met	
				National ISDN 1/2 Basic Access (C)	UCR Section 2.3.3	Met	
				Alerting Signals and Tones (R)	UCR Section 5.5	Partially Met	See notes 6 and 8.
				S/T Reference Point (R)	UCR Section 5.7.1.2.1	Met	
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
			VTC	ITU-T H.320 (R: BRI only)	FTR 1080B-2002	Met	
2-Wire Proprietary Digital	No	Not Certified (See note 2.)	Access	Directory Number Identification (R)	UCR Section 2.1.1	Not Tested	See note 2.
				Alerting Signals and Tones (R)	UCR Section 5.5	Not Tested	See note 2.
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	See note 2.

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Features and Capabilities						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
Common Features	Yes	Certified	Individual Lines (R)	UCR Section 2.1	Met	
			Selective call rejection (C)	UCR Section 2.1.2	Met	
			Denied originating service (C)	UCR Section 2.1.3	Met	See note 2.
			Code restriction and diversion (R)	UCR Section 2.1.4	Met	
			Call waiting (R)	UCR Section 2.1.5	Met	See note 9.
			Three-way calling (R)	UCR Section 2.1.6	Met	See note 10.
			Add-on transfer, conference calling, and call hold (C)	UCR Section 2.1.7	Met	
			Call Transfer Individual – All calls (R)	UCR Section 2.1.7.1	Met	See note 10.
			Call Transfer - Internal Only (R)	UCR Section 2.1.7.2	Met	
			Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R)	UCR Section 2.1.7.3	Met	
			Call Transfer – Outside (R)	UCR Section 2.1.7.4	Met	
			Call Transfer – Add-On Restricted Station (C)	UCR Section 2.1.7.5	Met	
			Call Transfer – Attendant (C)	UCR Section 2.1.7.6	Met	
			Call Hold (R)	UCR Section 2.1.7.7	Met	See note 10.
			Conference Calling – Six Way Station Controlled (C)	UCR Section 2.1.7.8	Met	
			Call Forwarding Variable (R)	UCR Section 2.1.8.1	Partially Met	See notes 10, 11, and 12.
			Call Forward Busy Line (R)	UCR Section 2.1.8.2	Met	See note 10.
			Call Forwarding – Don't Answer – All Calls (R)	UCR Section 2.1.8.3	Met	See note 10.
			Selective Call Forwarding (C)	UCR Section 2.1.8.4	Not Tested	See note 2.
			Call pick-up (C)	UCR Section 2.1.9	Met	See note 13.
			Address Translation (C)	UCR Section 2.7	Met	
			Assured Dial Tone (R)	UCR Section 2.9	Met	
Attendant	No	Not Tested	Attendant Features (C)	UCR Section 2.2	Not Tested	See note 2.
Public Safety	Yes	Certified	Emergency Service (911) Caller (R)	UCR Section 2.4.1.1	Met	
			Emergency Service (911) Public Safety Answering Point (C)	UCR Section 2.4.1.2	Not Tested	See note 2.
			Enhanced Emergency Service (E911) (C)	UCR Section 2.4.1.3	Met	
			Trace of terminating calls (R)	UCR Section 2.4.2	Met	
			Outgoing call trace (R)	UCR Section 2.4.3	Met	
			Tandem call trace (R)	UCR Section 2.4.4	Met	
			Trace of a call in progress (R)	UCR Section 2.4.5	Met	

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Features and Capabilities						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
Conferencing	Yes	Certified	Preset Conferencing (C)	UCR Section 2.6	Met	See note 14.
			Meet-Me Conferencing (R)	UCR Section 2.6.2	Met	See note 14.
			Progressive Conferencing (C)	UCR Section 2.6.3	Met	See note 14.
Nailed-up	No	Not Tested	Nailed-Up Connections (C)	UCR Section 2.8	Not Tested	See note 2.
DSN Hotline Services	Yes	Certified	DSN Analog Hotline Service (R)	UCR Section 2.12	Partially Met	See note 15.
MLPP	Yes	Certified	MLPP Overview (R)	UCR Section 3.1	Met	See note 16.
			Preemption in the Network (R)	UCR Section 3.2	Met	
			Network Facility with Lower Precedence Calls (R)	UCR Section 3.2.1	Met	
			Cancel to / Cancel from (C)	UCR Section 3.2.1.1	Not Tested	See note 2.
			Network Facility with Equal or Higher Precedence Calls (R)	UCR Section 3.2.2	Met	
			MLPP Trunk Selection (R)	UCR Section 3.2.3	Met	
			Hunt Sequence for Trunks (R)	UCR Section 3.2.3.1	Met	
			ROUTINE Precedence Calls (R)	UCR Section 3.2.3.1.1	Met	
			Precedence Calls Above ROUTINE Precedence (R)	UCR Section 3.2.3.1.2	Met	
			Method 1 (R)	UCR Section 3.2.3.1.2.1	Met	See note 17.
			Method 2 (C)	UCR Section 3.2.3.1.2.2	Met	
			MLPP Interworking with Other Networks (R)	UCR Section 3.2.4	Met	
			Precedence Call Diversion (R)	UCR Section 3.3	Met	
			Channel Associated Signaling (R)	UCR Section 3.4.1	Met	
			Primary Rate Interface (R)	UCR Section 3.4.2	Met	
			Analog Line MLPP (R)	UCR Section 3.5	Met	
			ISDN MLPP Basic Rate Interface (R)	UCR Section 3.6.1	Met	
			Single B Channel, Single Appearance, Single DN (R)	UCR Section 3.6.2	Met	
			Line Active with a Lower Precedence Call (R)	UCR Section 3.6.2.1	Met	
			Line Active with a Equal or Higher Precedence Call (R)	UCR Section 3.6.2.2	Met	
			Single B Channel, Multiple Appearances, Single DN (C)	UCR Section 3.6.3	Met	
			Two B Channels, Multiple Appearances, Single DN (C)	UCR Section 3.6.4	Not Tested	See note 2.
			Two B Channel, Two DN (Data Mode Only) (R)	UCR Section 3.6.5	Met	
			ISDN Primary Rate Interface (R)	UCR Section 3.7	Met	
			Precedence Call Waiting (R)	UCR Section 3.8.1	Met	
			Call Forwarding (R)	UCR Section 3.8.2	Met	
			Call Transfer (R)	UCR Section 3.8.3	Met	
			Call Hold (R)	UCR Section 3.8.4	Met	See note 18.
			Three-Way Calling (R)	UCR Section 3.8.5	Met	See note 19.
			Call Pickup (C)	UCR Section 3.8.6	Met	See note 13.

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Features and Capabilities						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
MLPP (continued)	Yes	Certified	Conferencing (C)	UCR Section 3.8.7	Met	
			Multiline Hunt Group (C)	UCR Section 3.8.8	Met	See note 10.
			Community of Interest (R)	UCR Section 3.8.9	Not Tested	See note 20.
			MLPP Common Channel Signaling Number 7 (C)	UCR Section 3.9	Met	
			CAS to CCS Trunk Network in a Mixed Media Network (C)	UCR Section 3.10	Met	
			MLPP Interaction with EKTS features (C)	UCR Section 3.11	Not Tested	See note 2.
Call Processing	Yes	Certified	Call Treatments (R)	UCR Section 4.1	Met	See note 21.
			Primary and Alternate Routing (R)	UCR Section 4.2	Met	
			E&M Lead Signaling States (C)	UCR Section 4.3.1	Met	
			4-Wire Analog User Access Lines (C)	UCR Section 4.3.2	Met	
			2-Wire User Access Lines (R)	UCR Section 4.3.3	Met	
			Termination of Analog Lines (R)	UCR Section 4.3.4	Met	
			DSN Interswitch Trunk Call Processing (non-CCS/ISDN) (R)	UCR Section 4.4	Met	
			DSN User Dialing (R)	UCR Section 4.5.1.1	Met	
			Interswitch and Intraswitch Dialing (R)	UCR Section 4.5.1.2	Met	
			Seven-Digit Dialing (R)	UCR Section 4.5.1.2.1	Met	
			Ten-Digit Dialing (R)	UCR Section 4.5.1.2.2	Met	
			Access Code (R)	UCR Section 4.5.1.3	Met	
			Access Digit (R)	UCR Section 4.5.1.3.1	Met	
			Precedence Digit (R)	UCR Section 4.5.1.3.2	Met	
			Service Digit (R)	UCR Section 4.5.1.3.3	Met	
			Route Code (R)	UCR Section 4.5.1.4	Met	
			Area Code (R)	UCR Section 4.5.1.5	Met	
			Switch Code (R)	UCR Section 4.5.1.6	Met	
			Line Number (R)	UCR Section 4.5.1.7	Met	
			Calling Name Delivery (C)	UCR Section 4.5.1.8.1	Not Tested	See note 2.
			Calling Number Delivery (R)	UCR Section 4.5.1.8.2	Met	See note 20.
			Emergency Service 911 Conflict Resolution (R)	UCR Section 4.5.1.9	Met	
			DSN Switch Outpulsing Digit Formats (C)	UCR Section 4.5.2	Met	
			Standard Directory Number (R)	UCR Section 4.5.3	Met	
			Standard Test Numbers (C)	UCR Section 4.5.4	Not Tested	See note 2.
			Base Services – Abbreviated Numbers (R)	UCR Section 4.5.5	Met	
			Digit Reception Requirements (R)	UCR Section 4.5.6	Met	
			Digit Registration Capacity (R)	UCR Section 4.5.7	Met	
			Screening (R)	UCR Section 4.5.8	Met	

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Features and Capabilities						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
Network Management	Yes	Certified	Interfaces (R)	UCR Section 9.1	Met	See note 22.
			Data Quality (R)	UCR Section 9.2.1	Met	
			Traffic Measurements (R)	UCR Section 9.2.2.1.1	Met	
			Reference Data (C)	UCR Section 9.2.2.1.2	Not Tested	See note 2.
			Line Servicing (C)	UCR Section 9.2.2.2	Not Tested	See note 2.
			Trunk Groups (C)	UCR Section 9.2.2.3	Not Tested	See note 2.
			Call Processors (C)	UCR Section 9.2.2.4	Not Tested	See note 2.
			Switch Services (C)	UCR Section 9.2.2.5	Not Tested	See note 2.
			Special Studies (C)	UCR Section 9.2.2.6	Not Tested	See note 2.
			Remote Switching Studies (C)	UCR Section 9.2.2.7	Not Tested	See note 2.
			Features (C)	UCR Section 9.2.2.8	Not Tested	See note 2.
			Common Channel Signaling Network Measurements (C)	UCR Section 9.2.3	Not Tested	See note 2.
			ISDN Measurements (C)	UCR Section 9.2.4	Not Tested	See note 2.
			Traffic Capacity (R)	UCR Section 9.2.5	Met	
			Fault management (R)	UCR Section 9.3	Met	
			Configuration management (R)	UCR Section 9.4	Met	
			Call Detail Recording Data Retention (C)	UCR Section 9.5.2	Not Tested	See note 2.
			Performance management (R)	UCR Section 9.6	Met	
			Network Management controls (C)	UCR Section 9.7	Not Tested	See note 2.
			Remote access (R)	UCR Section 9.8	Met	
ISDN Services	Yes	Certified	BRI Access, Call Control and Signaling (R)	UCR Section 10, Table 10-1	Met	
			Uniform Interface Configuration for BRIs (R)	UCR Section 10, Table 10-2	Met	
			Electronic Key Telephone Systems (EKTS) (C)	UCR Section 10, Table 10-3	Not Tested	See note 2.
			PRI Access, Call Control and Signaling (R)	UCR Section 10, Table 10-4	Met	
			PRI Features (R)	UCR Section 10, Table 10-5	Met	
			Packet Data Features and Capabilities (C)	UCR Section 10, Table 10-6	Not Tested	See note 2.
Synchroniz-ation	Yes	Certified	External Timing Mode (C)	UCR Section 11.1.1.1	Not Tested	See note 2.
			Line timing mode (R)	UCR Section 11.1.1.2	Met	
			General (C)	UCR Section 11.1.2.1	Not Tested	See note 2.
			Internal Stratum 4 (R)	UCR Section 11.1.2.2	Met	
			Synchronization Performance Monitoring Criteria (C)	UCR Section 11.2	Not Tested	See note 2.
			DS1 Traffic Interfaces (C)	UCR Section 11.3	Met	
			DS0 Traffic Interconnects (C)	UCR Section 11.4	Not Tested	See note 2.

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Features and Capabilities						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
Reliability	Yes	Certified	Reliability Requirements (R)	UCR Section 12.1	Met	
			Backup Power (R)	UCR Section 12.3	Not Tested	See note 23.
			Power Components (R)	UCR Section 12.3.1	Not Tested	See note 23.
			UPS Requirements (R)	UCR Section 12.3.2	Not Tested	See note 23.
			UPS Load Capacity (R)	UCR Section 12.3.2.1	Not Tested	See note 23.
			Backup Power (Environmental) (R)	UCR Section 12.3.3	Not Tested	See note 23.
			Alarms (R)	UCR Section 12.3.4	Not Tested	See note 23.
Security	Yes	Certified	GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	UCR Section 13	Met	See note 24.
VoIP						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
VoIP System	No	Certified	Voice Quality with MOS of 4.0 or better (R)	UCR App. 3, para. A3.2.1	Met	
			ITU-T G.711 PCM CODEC (R)	UCR App. 3, para. A3.2.2	Met	
			MLPP (R)	UCR App. 3, para. A3.2.3	Met	
			Security (R)	UCR App. 3, para. A3.2.4	Met	
			Network management (C)	UCR App. 3, para. A3.2.5	Met	
			System timing (R)	UCR App. 3, para. A3.2.6	Met	
			Latency ≤ 60 milliseconds (R)	UCR App. 3, para. A3.2.7	Met	
			IPv6 capable (R)	UCR App. 3, para. A3.2.8	Not Tested	See notes 25 and 26.
			Service Class Tagging (R)	UCR App. 3, para. A3.2.9	Met	
			VoIP System Downtime (IP network 35 min/yr Subscriber 12 min/yr) (R)	UCR App. 3, para. A3.2.10	Met	
Network Gateways						
Gateway	**Critical**	**Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
PSTN (See note 27.)	No	Certified	Trunking: Positive Identification Control (C)	CJCSI 6215.01C	Met	
			Trunking: On-Netting (C)	CJCSI 6215.01C	Met	
			Trunking: Off-Netting (C)	CJCSI 6215.01C	Met	
			Trunking: Ground Start Line (R)	UCR Section 5.2.2	Met	
			Trunking: Immediate Start (C)	UCR Section 5.3.2	Met	
			Trunking: Delay Dial (C)	UCR Section 5.3.4	Met	

Table 2-4. SUT Interoperability Requirements/Status (continued)

NOTES:

1 Met all critical CRs and FRs with the REDCOM enclave. Although Cisco offers this interface, it is not certified.

2 This feature is not supported by the SUT. This is not a required feature for a SMEO. There is no risk associated with the SUT not supporting this feature.

3 The SUT does not support NFAS on the Cisco enclave ISDN PRI National ISDN Standard 2 (NI2) interface. NFAS is supported on the REDCOM HDX. Therefore, the REDCOM enclave must include at least one HDX shelf equipped with multiple ISDN PRI T1s. Both enclaves do support FAS.

4 A discrepancy exists that is associated with the monitoring tool that Cisco enclave uses to check the status of the ISDN PRI trunks on the gateway. The monitoring tool occasionally provides an inaccurate representation of the status of the channels on the trunks when they are busied by the remote switching system. The SUT will occasionally provide an indication that the channel that was busied out by the far-end switch remains in an idle condition. This anomaly can be eliminated by insuring the trunks are busied at both the remote end and at the SUT. Furthermore, when this anomaly does occur, the correct busy state of the trunks is reflected in layer 3 protocol of the ISDN PRI interface, therefore, the operational impact is minor.

5 Met all critical CRs and FRs with the REDCOM enclave. The E1 ISDN PRI interface is supported by the Cisco enclave; however, it does not support ITU-T Q.955.3 MLPP.

6 The conference disconnect tone that is provided by the REDCOM enclave does not meet the specifications designated in UCR, section 5.5.2. The SUT conference disconnect tone is distinguishable from other DSN tones and cadences; therefore, this anomaly has a minor operational impact.

7 A configuration change was required on the Cisco enclave analog gateways to meet the requirement for interoperability with secure devices, specifically the L3 Omni Secure Wireline Terminal. On the individual voice ports, the minimum and maximum settings for "timing hookflash in" had to be changed to a maximum value of 500 ms and a minimum value of 150 ms. Otherwise, a call that is placed between two Omni devices on the SUT will not disconnect when placed on hook.

8 The precedence above ROUTINE ringing cadence that the REDCOM enclave applies to BRI phones does not meet the specifications as detailed in the UCR, section 5.5.1. The precedence above ROUTINE cadence is distinct from the ROUTINE cadence when it is configured properly; therefore this anomaly has no operational impact.

9 The Cisco enclave does not support Call Waiting. However, there is no operational impact because the requirement is satisfied with multiple line appearances having a busy trigger. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

10 Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog stations of the Cisco enclave. Call Forward Variable, Three-way Calling, Call Hold, and Call Transfer are supported on VoIP stations only. These features are required for a SMEO for all instruments; however, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

11 When CFV is assigned to any station on the REDCOM enclave and CFV is invoked by the user, any station with CFV invoked does not receive a “ping” ring when calls are being forwarded. The operational impact is minor.

12 A short "ping" ring is not provided when calls are forwarded on the Cisco enclave; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact.

13 The SUT does not support Call Pickup between the Cisco enclave and the REDCOM enclave. This solution is unique in that it offers SMEO functionality with two switches (Cisco and REDCOM) and each switch offers call pickup. This action is mitigated by not mixing call pickup groups between the two switch enclaves. DISA’s adjudication of this discrepancy was ruled to have a minor operational impact.

14 Met all critical CRs and FRs with the REDCOM enclave only.

15 The SUT does not support the ISDN PRI code set 5 off hook indicator elements for hotline services as required by the UCR. The vendor began testing prior to 14 June 2008 and, therefore, was not required to provide this feature. This anomaly has minor operational impact.

16 The SUT does not support the Loss of C2 announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. This requirement is currently under review by DISA and the Joint Staff. In addition, the specific conditions that invoke this announcement have not yet been defined. As a result, the vendors are not required to be in compliance until 18 months from the date the requirement is fully defined.

17 The SUT does not support Method 1 preemption search algorithm if the trunks are a combination of the Cisco and REDCOM enclaves. In order to use the Method 1 search preemption search algorithm, all trunk groups must be member of the Cisco Gateway or the HDX switch. DISA’s adjudication of this discrepancy was ruled to have a minor operational impact.

18 The SUT does not maintain the precedence level when transferring a call between the Cisco enclave and the REDCOM enclave. This discrepancy is due to the functionality between the Cisco and REDCOM enclaves. DISA’s adjudication of this discrepancy was ruled to have a minor operational impact.

19 When the initiator of a three-way call is preempted, the remaining parties do not receive a conference disconnect tone. However, the remaining members of the three-way call do stay connected. DISA’s adjudication of this discrepancy was ruled to have a minor operational impact.

20 This is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

Table 2-4. SUT Interoperability Requirements/Status (continued)

NOTES continued:

21 The SUT does not support the full complement of CoS tables as specified in the UCR. The SUT supports 255 CoS tables for analog lines and does not support CoS tables on access lines, number codes, trunks, or groups of trunks. This limitation has posed a minor operational impact within the DSN when assigning lines and trunks on the SUT.

22 Met all critical CRs and FRs with an IP interface.

23 This requirement is a non-testable requirement. It is the responsibility of the respective base/post/camp/station communication agency to provide this with the SUT when installed.

24 Security is tested by DISA-led Information Assurance test teams and published in a separate report, reference (c).

25 An IPv6 capable system or product, as defined in the UCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor LoC signed by the Vice President of their respective company. The vendor stated in writing, their intent to return to JITC for testing of their solution with IPv6 enabled earliest date available. In addition they stated in writing, compliance to the following criteria:

a. Conformant with IPv6 standards profile contained in the DISR. These standards are delineated in the UCR, appendix 11.
b. Maintaining interoperability in heterogeneous environments and with IPv4.
c. Commitment to upgrade as the IPv6 standard evolves.
d. Availability of contractor/vendor IPv6 technical support.

26 The SUT was tested with IPv4 only. In accordance with the Office of Secretary IPv6 Rules of engagement a solution can be tested and certified for IPv4 only, however the vendor is required to stipulate in an IPv6 LoC their way ahead to be IPv6 capable by end of CY 2008. In addition the vendor is required to return for retest with this IPv6 solution prior to the end of CY 2008. The vendor stated in their IPv6 LoC submission that they will not be able to deliver an IPv6 capable solution until 31 May of 2010. The vendor received a waiver for this requirement from OSD on 9 March 2009.

27 Voice, facsimile, data, and VTC service requirement s for PSTN are identical to DSN with the exception of MLPP.

Table 2-4. SUT Interoperability Requirements/Status (continued)

LEGEND:

ANSI	American National Standards Institute	FTR 1080B-2002	Video Teleconferencing Services	para.	paragraph
App.	Appendix	G.711	PCM of voice frequencies	PBX	Private Branch Exchange
BER	Bit Error Ratio	GR	Generic Requirement	PCM	Pulse Code Modulation
BRI	Basic Rate Interface	GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security	PCM-24	Pulse Code Modulation - 24 Channels
C	Conditional			PCM-30	Pulse Code Modulation - 30 Channels
C2	Command and Control			PMO	Program Management Office
CAS	Channel Associated Signaling	H.320	Standard for Narrowband VTC	PRI	Primary Rate Interface
CCS	Common Channel Signaling	HDX	High Density Exchange	PSTN	Public Switched Telephone Network
CFV	Call Forwarding Variable	IP	Internet Protocol	Q.955.3	ISDN Signaling Standard for E1 MLPP
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	IPv4	Internet Protocol version 4	R	Required
CODEC	coder/decoder	IPv6	Internet Protocol version 6	S/T	ISDN BRI four-wire interface
CoS	Class of Service	ISDN	Integrated Services Digital Network	SMEO	Small End Office
CRs	Capability Requirements	IT	Information Technology	SS7	Signaling System 7
CY	Calendar Year	ITU-T	International Telecommunication Union-Telecommunication Standardization Sector	STE	Secure Terminal Equipment
DIACAP	DoD Information Assurance Certification and Accreditation Process			STIGs	Security Technical Implementation Guides
DISA	Defense Information Systems Agency	JITC	Joint Interoperability Test Command	STU-III	Secure Telephone Unit -3rd generation
DISR	DoD IT Standards Registry	kbps	kilobits per second	SUT	System Under Test
DN	Directory Number	LoC	Letters of Compliance	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DoD	Department of Defense	Mbps	Megabits per second		
DoDI	DoD Instruction	MFR1	Multi-Frequency Recommendation 1	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DP	Dial Pulse	min	minute		
DS0	Digital Signal Level 0 (64 kbps)	MLPP	Multi-Level Precedence and Preemption	T.4	Standardization of Group 3 facsimile terminals for document transmission
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	MOS	Mean Opinion Score		
		ms	milliseconds	UCR	Unified Capabilities Requirements
DSN	Defense Switched Network	NFAS	Non-Facility Associated Signaling	UPS	Uninterruptible Power Supply
DTMF	Dual Tone Multi-Frequency	NI 1/2	National ISDN Standard 1 or 2	VBD	Variable bit data
E&M	Ear and Mouth	NX56	Data format restricted to multiples of 56 kbps	VTC	Video Teleconferencing
E1	European Basic Multiplex Rate (2.048 Mbps)			VoIP	Voice over Internet Protocol
EKTS	Electronic Key Telephone System	NX64	Data format restricted to multiples of 64 kbps	yr	year
FRs	Feature Requirements				
FTR	Federal Telecommunications Recommendation	OSD	Office of the Secretary of Defense		